# Synthetica

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# Contents

# Chapter 1

# Synthetica

## 1.1 Aural Synthetica v1.1 Manual © Blachford Technology 28/11/95

Aural Synthetica main Manual v1.1

by Nicholas Blachford 16/4/96

If you have read through the Startup Guide/Tutorial you should now

have a fairly good idea of how this program works, this manual goes

into more detail on what the individual sliders and buttons do but

it is not a tutorial.

## 1.2 Aural Synthetica Version 1.1 © Blachford Technology 1995-96

Welcome to Aural Synthetica v1.1

Aural Synthetica is a sound generation program based upon a versatile modular synthesizer. The sounds are generated by large numbers of calculations which produce a 16 bit sound sample in memory, the sound is not produced in real time. This sample can then be saved in IFF, AIFF, WAV, MAUD and SAFF formats.

What is a Modular Synthesizer?

Most synthesizers are based around certain elements which are organised in a certain order to generate their sound, these elements consist of an Oscillator which generates a fixed sound, a filter which cuts or boosts different frequencies and an amplifier which determines the volume of the sound, an Envelope generator is used to control the amplifier changing the volume of the sound over time, Another element also included is a Low frequency Oscillator which can be used to change the frequency over time giving a vibrato sound.

This is more or less the way synthesizers have been organised since the early 1970's. Previous to the 70's synthesizers had the same elements but they were not fixed in any way. The elements were called modules and could be connected using patch chords (wires) in any way the user chose, this gave rise to a range of sounds way beyond the scope of normal synthesizers with their fixed design.

Aural Synthetica is a modular synthesizer modelled in software but using digital technology to add many more modules and module types than would have been found in a 1960's modular Synth.

A comprehensive Patch programmer allows the user to arrange the modules in any way they choose using software based patch chords to link modules. Patches can then be loaded, saved and edited at will. There are also 30 preset "Basic Synths" which consist of preset patch setups, some of these allow one of the modules to be changed by pressing a single button, in addition to this all the module settings can still be changed allowing a very variety of sounds.

To generate sounds the Oscillators use either the 12 basic wave

forms or the user can create 24 of their own their own. A comprehensive waveform editor gives the user the tools to create an almost infinite number of waves.
There are also 8 types of Module all of which can be used to process signals in one or more ways, these also allow connections between different modules to be made.

## 1.3 The Main Screen

The Main Screen
The Main Screen consists of two Windows, The Digital Modular Synthesizer (DMS) window and the View Samples window. Upon bootup the program also displays a window allowing the user to select the amount of memory to be taken by the program, If Virtual memory is available the program will also allow this to be chosen. Also upon bootup the program switches the Audio filter off and allocates Audio channels if it can, if it cannot allocate Audio channels the program runs without them allowing another Audio program to be run concurrently, however in this case the Play button is ghosted so you can't press it.

## 1.4 View Samples Window

VIEW SAMPLES Window
The View Samples window displays the sample when created and allows to see to view samples in Mono, Stereo, and both stereo sides, In mono mode only a mono display can be seen and the selection buttons are disabled.
Play - This plays the contents of the sample in memory, The Amiga is not normally capable of playing samples at 44.1KHz so the sample is played at 22KHz missing every other sample, this can make some samples with a lot of high frequency content sound slightly different sometimes than if played with a proper 44.1KHz system, 8 bit audio playback also reduces the quality of samples especially in the bass region, some bass samples will little or no high/mid frequency content can sound very noisy on 8 bit playback.
When you press play the entire sample is played whether you like it or not, once you press play it only stops once it gets to the end of the sample.

## 1.5 Pull Down Menus

PULL DOWN MENUs

The Pull Down menus can only be accessed while the View window

is active, there are two main menus PROJECT and EDIT.

PROJECT menu

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New All - This resets the entire program, it deletes all patch

connections and resets the Module settings to their defaults.

New Patch - This has no effect on the Module settings but it deletes

all patch connections.

Save Sample - This allows you to save your created sample in one of

5 file formats. See File Formats for more details.

NOTE: When saving a Sample or Patch you MUST first select the drawer

to save it in - even if it is the default, if you don't select a

drawer the program will save nothing. If in doubt simply put your

creation in the Samples or Patch drawer.

Load Patch - This loads a presaved Patch setup, this includes all

patch connections and module settings. A number of Patches have

been supplied in the Patch drawer.

Save Patch - This saves all Patch connections and module settings

in a single file, The user created waveforms are not saved in this

option.

Save Patch + Waves - This is similar to Save Patch but this time any

user created waves are also saved. These files are bigger as each wave

saved uses 2K memory space.

About Aural Synthetica - This displays some information about the

program.

Quit - This opens a window asking if you wish to quit and allowing

you to do so.

EDIT Menu

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Zero Sample - This deletes the entire contents of the sample

memory.

Toggle Filter - This allows the Audio filter to be toggled.

## 1.6 File Formats

Aural Synthetica supports the saving of 5 file formats,
none of these are compressed and all samples are saved at a
sampling rate of 44190 Hz.
SAFF - The native format introduced in Syntheticas sister program
Aural illusion, samples are saved as 16 bit data.
for more details see the SAFF Specification.
AIFF - This is a 16 bit file format used on the Amiga and Apple
Machintosh, It also turns up on the PC as .AIF, If you have a
Aura 12 bit sampler the Aura software uses AIFF for it's 16 bit data.
WAV - This is a format used mostly on the PC, this is used in Windows
for storing audio. If you are porting samples onto the PC this is
the one to use, samples are saved as 16 bit data.
IFF - This is the Amigas 8 bit file format, most Amiga programs
support this format but it is only 8 bit and quality is thus
somewhat lower.
MAUD - If you have a Wavetools sound card for the Amiga you will want
this, the Samplitude software and some other Amiga audio programs
support it, data is 16 bit.
Once saved the samples you have created may need a touch of editing to
remove clicks or other glitches which tend to appear at the beginning
and end of Synthetica samples, this problem has been reduced in v1.1 but
can still occur. Any sample editor can remove these noises.

## 1.7 SAFF Specification

Simple Audio File Format Specification
Revision 1.2 6/12/95
NOTE: Aural illusion v2.0 and Aural Synthetica both support the SAFF
format but only use a maximum of 2 channels, so there is no need to
support the block interleave feature at present.
A Compressed SAFF format is planned for the future which will use
lossless compression.
I have added a version number to the header, this
will indicate if compression or block interleave is present as well as
specifing the maximum number of channels, See below for more details.
---- SAFF Specification ----
An SAFF (Simple Audio File Format) file consists of a very small 32 Byte

header file which holds the basic sample info followed by the samples itself, the samples can be any size you want and can contain up to 255 channels. The samples can be as many bits as you want but the data is stored in minimum units of 8 (1 Byte) You can store 8, 16, 24, 32 or more bits per sample right the way up to 2040 bits (255 bytes)! Aural illusion v2.0 however does not support anything over 16 bit as yet.

You can have up to 255 channels worth of data, these are not interleaved like the other formats but are stored one after the other, the reason for this is speed, it is much faster to load in the sample data in single lumps without having to de-interleave them, it also means you don't have to write a different stereo loading routine - you just use the mono one twice.

The data can be interleaved in blocks of 25000 samples if the Block Interleave Byte is set to 3, in this case 25000 samples of channel 1 are saved followed by 25000 samples of channel 2 then back to channel 1 etc... This will allow data to be played from a hard disc with multiple tracks all in one file but without having to search or de-interleave data. Aural illusion v2.0 ignores the Block interleave byte at the moment.

SAFF also allows you to store your sample data in Least Significant Bit first format, this means sample data could be transferred to and from the PC without difficulty, by default data is stored in Most Significant Bit first order. (Amiga / Mac / ST standard).

The header as well as being small is also easy to decode. All Longs are on Long offsets and the same applies for the sample Words/Bytes.

---- The SAFF Header ----

The SAFF header is at the beginning of a SAFF file and is 32 Bytes long.

Offset Bytes Description Type

0 4 Ascii code - "SAFF" Ascii SAFF

4 1 Channels Byte 1 or more

5 1 Bytes Byte 1 or more

6 1 Loop Byte 1=on 0=off

7 1 Byte Order Byte 0=MSB 1=LSB

8 4 Sample Length ULONG Samples per channel

12 4 Playback Rate ULONG Samples per second

16 4 Loop Start ULONG Offset from 0

20 4 Loop End ULONG Offset from 0

25 1 MIDI Byte 0=off 1=on

26 1 MIDI NOTE Byte Note number (0-127)

27 1 Compression type Byte 0=none See below

28 1 Block interleave Byte 0=none See below

29 1 SAFF Version Byte 0=standard See Below

30 1 Reserved Byte 0

31 1 Reserved Byte 0

32 1 Extra Header Byte 0=none 1=more data

ULONG = 32 bit unsigned

Byte = 8 bit signed

The sample length is in samples NOT words, to calculate the size of the sample data multiply the number of samples by the number of bytes.

For the entire samples multiply the above resulting number by the number of channels.

There are no standard compression types as yet but I may add some in the future, so the compression byte should be set to zero for now.

Block interleave sizes (in samples - NOT Bytes or Words)

0=none

1=10000

2=15000

3=25000

4=50000

5=75000

6=100000

---- The sample Data ----

Data is saved as signed words, be they 8, 16, 24, 32 or however many bits, if your data is held as unsigned words convert them into signed words as you save them. If you are storing 12 or 14 bit data store it as 2 bytes.

A 24 bit sample will be stored as 3 bytes.

Data is stored in sequential blocks starting with the first channel.

The sample data is not interleaved so loading and saving can be very fast achieved with memory dumps.

Block Interleave

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Block interleave allows the interleaving of channels in blocks, 25000 samples at a time, this will allow you to read or write more than one channel at a time direct from Hard disc, The blocks are stored :

Channel 1,2,3,4,1,2,3,4 etc...

If you were using a block size of 25000 (Block interleave=3) and 2 tracks of 16 bit data you would read 50000 bytes of channel 1 then 50000 bytes of channel 2 then 50000 bytes of channel 1 and so on...

By reading 100000 bytes at a time and setting the hardware audio pointers to the beginning of each block then using double buffering you could eaisly

play multitrack audio from a hard disc without using almost any CPU power.
It is unlikely that the sample you are saving will have a size that is an exact multiple of the Block interleave size, so the final blocks should be shortened to allow for this, padding out with zeros would be a waste of space so this method is not used, file readers should take account of this.

LSB data

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If you are using a PC and wish to store data in SAFF format set the LSB byte to 1 so other platforms know the data bytes will need to be reversed.
If when reading the file check the first 4 bytes ASCII code, they should read as "SAFF" but if they read ASFF the entire file will need to be byte swapped.

Versions of SAFF

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The Version number has been added to avoid a little confusion,
If you want to add a reader to you program you would have to support the features even if you don't expect or require them, i.e. Aural illusion and Aural Synthetica can only use a maximum of 2 channels and thus Block interleave is fairly useless so adding a version number helps out here, different versions of SAFFs will support features, more versions may be added at a later date. See the following table for details:

| Version | Block Int | Compression | Max Channels | Max Bytes per sample |
| --- | --- | --- | --- | --- |
| 0 | No | No | 2 | 2 |
| 1 | No | Yes | 2 | 2 |
| 2 | Yes | No | 255 | 2 |

If you are adding SAFF support (go on - it's easy!) only support the Versions you need, if you find a different version simply tell the user it's the wrong type, Aural illusion and Aural Synthetica both only support Version 0 SAFFs.

---- Copyright & future changes ----

Write zero into the final four bytes as these are reserved for future use, perhaps to indicate another header before the sample data. If more is added they will be in the form of more 32 byte headers, a zero value in the last ULONG will indicate sample data is next.
I will retain the copyright of the SAFF format but only to stop it getting messed up and confused. Anyone can use the SAFF format if they wish, just don't change anything. It's designed as a simple format for storing and transferring sample data, if you wish to add to the specification contact

me FIRST and I can update the format specification. There are no charges for using SAFF and never will be - it's making for transfer of samples easier, not profit.
Some public domain programs will be release at a later date to deal with SAFFs, these will probably be Amiga only but I may do other platforms as well.


## 1.8 The Digital Modular Synthesizer (DMS) Window

DIGITAL MODULAR SYNTHESIZER (DMS) window
This Window is where all the other sample control windows can be accessed from. The bottom of the window contains a message box for displaying various messages. Beside this are the buttons for accessing the Wave Editor, Basic Synthesizers and Patch Programmer, at the end is a Render button which opens a window which starts the creation of a sample, the window allows you to select where your sample is to go, if you select one of the Left or Right options the sample will be rendered into that side, when one of the mono modes is used both outputs are summed.
Most of the DMS window is taken up by the buttons for accessing the modules themselves, there are 66 modules in all and there is a but ton for each one of them. There is a division between the signal Generation modules and the Processing modules, this is simply to indicate the difference between them, both the Oscillators and Envelopes can generate signals but the Envelopes have a modulation in put section so it can also be used to process signals, Oscillators on the other hand have no input but can be modulated in four different ways.
The Modules in the right hand block are for processing signals, they accept a main input and can then process it in a number of ways, there are one or more modulation inputs which take a signal and use it to control the signal processing within the module. On the far right are the Tool and Left/Right modules, the Left/Right modules are the main output modules and the samples created with the program are taken from these modules and therefore one of them must be connected when a sound is created.

## 1.9 The Waveform Editor

WAVEFORM EDITOR

The Waveform Editor is used to create waves which are used within the oscillators, these can be anything from simple variations on the 12 basic waves to complex shapes which look like nothing else.

Basic Waves

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There are 36 waves in Aural Synthetica, 24 of these can be edited and changed at will but the first 12 are fixed, waves 13 - 24 are blank and can be made by the user, waves 25-36 are not blank but can still be used by the user, they have waves in them as they are used to demonstrate the Waveform modulation by the Basic Synths 25 - 33, changing these wave shapes also alters the sound created by these Basic Synths and saves you having to set up the oscillators by hand.

There are two basic ways of making a waveform, this can be mixing waves together or mixing different harmonics, you can also mix both types. Once the waves are created you can also change the wave forms using a number of functions.

Before trying anything make sure that you have picked a wave to work on, this is done with the wave slider, you can use any wave from 13-36 for wave creation but not waves 1-12.

Wave Mixing

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The top right hand side of the window is occupied by 24 sliders each with a level, these represent the first 24 waves. There are no default values so pressing "Mix Waves" will create a blank wave, to mix the waves you must set the values of at least one slider then press Mix Waves. You can mix as many waves as you wish.

Harmonic waves

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You can also use Harmonic wave synthesis to create waves. A plain sine waveform will sound very dull, it can be brightened up by increasing the number of harmonics in it, a Harmonic is a copy of the same waveform but at a different frequency, the number of a harmonic relates to the difference between the frequency of it and the original wave, a second harmonic is double the frequency, a third triple and so on.

To make a sound more interesting it is possible to change the harmon

ics over time and you will that most of the processing effects within
Aural Synthetica will have some effect on the harmonics within a wave.
The Wave editor uses the sliders on the bottom right hand side to add
harmonics to a wave, the number below a slider tells you which harmonic
is represents, the wave below this is the base wave, this is used
to select which wave is used as the First (fundamental) harmonic,
changing this has a big effect on the sound since all but the first wave
already contain extra harmonics and creating extra harmonics of this
wave will create harmonics of these harmonics.
To create Harmonic waves set some of the sliders and press "Mix
Harms", the first slider is set to maximum by default but this can be
changed as you wish. If you wish to see the effect of adding harmonics
to waves look at waves 29 - 36, these start at a basic sine wave but
various are added until you get to wave 36 which is full of different
harmonics. Changing harmonics over time is used in Wave modulation in
the Oscillators.

Wave Combinations
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In addition to Adding waves or Harmonics it is possible to mix both
types at the same time, pressing "Mix Both" has this effect.
It is also possible to change the way waves are mixed using the radio
buttons and Phase slider, these can be used together with the other
methods.
The Phase slider sets the shift between each wave being added.
The Radio Buttons allow you to select a combination method,
Add - simply adds the waves together.
Sub - Subtracts the waves from one another.
Mul - Multiplies the waves together, do not use too many waves as
they can distort.
> - Takes the greatest value of the waves.
> - Takes the smallest value of the waves.
OR - OR the waves together.
XOR - XORs the waves together.

Modifying Waves
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There are a number of options to further change the waveforms you
have created, these require a wave to be present to work on.
Reverse - Reverses the direction of a wave.
Invert - This Turns a wave inside out, changing the Radio buttons

changes how this works.

Make Noise - Turns a waveform into noise.

Inc Noise - Adds a small amount of noise to a wave.

Random - Creates a new waveform from scratch, this does not require a wave to be present and will overwrite any there.

## 1.10 Basic Synthesizrs

BASIC SYNTHESIZERS

These are preset patches which create sounds and show you how the program works, in most cases they also allow you to change one of the modules used.

There are 51 Basic Synthesizers all of which have been set up to produce a sound of some sort, the sounds produced are designed to show what the program is capable of, the sounds may not be very musically useful but should be interesting and useful as a starting point to create new sounds. The Modules used in any Basic Synth are displayed but have to be selected from the DMS Window in the usual way, once selected all of the controls can be changed at will, this way you can change the sound of a Basic Synth into something completely different. It should be noted that if the Basic Synth window opened after patch changes have been made they will be deleted, some of the other settings may also change.

In addition to the Basic Synths there are Variations, these are Basic Synths where you can change one of the modules, If you select Basic Synth number 3 using the Synth slider it will be displayed along with a description, the module that can be changed is the one which appears raised more than the others. On the right of the Synth Slider is a row of 7 buttons with names of Modules on them, if Modf is pressed, You will notice that "Mix 1" in the raised box will change to "Modf 1", the row of buttons are used to change the raised modules in the Basic Synths and also the patch set up, if you select one of the buttons while the Patch Programmer is open it will automatically redraw with the new setup.

## 1.11 The Patch Programmer

THE PATCH PROGRAMMER
This is the Window which is full of little buttons, If you follow the
Startup Guide/Tutorial it will take you through the basics of using the
patch programmer.
The Programmer was designed to allow the user full access to all the
inputs and outputs in the system, the Programmer allows the user to
view and edit the current Patch.
Patching Controls
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Add/Cut - This selects between Adding or Cutting Patch leads.
To add a patch lead select a small (input) button followed by a larger
(Output) one or vice versa, you can only connect Inputs to Output or
Outputs to Inputs.
To Cut a patch lead select the Cut radio button, the Programmer will
then redraw itself and all the outputs will be ghosted, in Cut mode any
input selected which has a patch lead going to it will be disconnected,
the patch lead will be drawn over with gray, if the display becomes
messy or unreadable press Redraw.
Clear View - This redraws the programmer but does not include any
patch leads, the leads are still there but are not shown, this function
is to allow the programming of complex patches where some connections
may be unclear to see, pressing this allows more connections to
be added without complications.
Redraw - This redraws the patch programmer and any patch leads, if a
patch lead is cut within a module the lead will still be shown even
though it may not in fact be there, redrawing clears these.
New - This Clears all patch leads and redraws the Programmer, the
other module settings remain unaffected.
Test - Checks to see if the patch can compute, this is not a very
complex checking system so it is still possible that no sound will
be created.
Render - Starts Sample Creation.
Note: if you patch isn't properly set up it won't render and just give
you an "Errors in Patch" in the DMS message box.
Please remember to follow the patching rules (in the included startup
Guide/Tutorial).

## 1.12 Rendering a Sample

RENDERING A SAMPLE
Pressing any of the three Render buttons will cause the Rendering Options Window to open.
This window allows you to select which channel to render to:
Mono All - Uses all sample memory to create one big sample.
Mono Left - Creates a sample in the Left channel, (no effect on Right).
Mono Right - Creates a sample in the Right channel (no effect on Left).
Stereo - Creates a Stereo sample using both channels.
Quality - Picks between two different tuning algorithums, the Less option can create distorted waveforms at certain frequencies, these can be heard if you sweep the frequency of an oscillator. The distortion does not occur with the Good option.
Length - Allows to determine the length of the sample between 1 and 100% of allocated memory. If you are creating a complex sample it is best to do test renders using short sample lengths then create the full sample when you are sure it will be ok.
Instability - This can be used to create a detuned effect by adding random detunes to the oscillators, best kept below 5.
Cancel - Closes Rendering Options window.
Start Render - Starts the Computation
The Rendering Process
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When you press Start Rendering all except the View Samples and DMS windows close, a new window then reopens called Rendering Sample and a red bar will move across the window, this will most likely move rather slowly as Aural Synthetica involves some very heavy computation. When the computation is complete the Rendering Window will close and the resulting sample will be shown in the View Samples Window. This can then be played if sample channels have been allocated. When you press play the entire sample is played, it only stops playing when the end is reached, there is no way of stopping playback before.

## 1.13 Oscillators

OSCILLATORS
Oscillators are used to create sounds which are further processed in other modules. Oscillators can also be modulated themselves in four different ways and can thus create a very wide range of sounds, these modulation types can be used simultaneously creating very complex sound waveforms, these oscillators can be considered as miniature synthesizers in their own right.
The Oscillator window is split into 5 parts:
Waveform Modulation.
Main Oscillator Controls.
Phase Shift Modulation
Pulse Width Modulation.
Frequency Modulation.
Main Oscillator Controls
------------------------
These are located about half way down and consist of 6 Sliders and two check boxes:
Waveform - This sets which waveform is used by the oscillator to create the sound, this is the basis for the sound created by the oscillator as it is this wave which goes on to be modulated and shaped.
Amplitude - This determines the volume of the Waveform, this is preset at 256 which is the normal maximum level, above this distortion sets in and the sound changes.
Delay - For some sounds you may want a period of silence before they begin, this could be like strumming a guitar where the lower pitched strings are hit before the higher strings. The default is 0 but it goes up to 128, this produces a delay of approximately 1 second.
Note - This determines the note which is generated by the oscillator, this ranges between 0.06Hz to 3951Hz, If Frequency Modulation is used this range can increase.
Octave - This is used with the note control to change the output frequency, changing the octave either doubles or halves the frequency.
Detune - To thicken up sounds multiple Oscillators are used all slightly detuned, this slider slightly changes the frequency.
NEG - This turns the output of the Oscillator upsidedown.
REV - This reverses the output of the Oscillator.

Waveform Modulation

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This form of modulation changes the shape of a waveform over time, to use it the waveform must be set to 0, the waveform sliders along the top of the window set and a modulation source selected.

Below the 8 wave displays are the Wave Modulation controls:

NEG - This flips the modulating waveform upside down.

Source - This selects the input for modulation, it is this input which selects the waveform from the 8 waves at the top.

Copy - This copies the waveform setup from another oscillator, the slider below picks the oscillator.

Reverse - This Reverses the 8 waveforms.

Used on it's own (especially without detuning) waveform modulation creates sounds which are totally different to anything else created with Synthetica, the nearest thing sound wise would be if you have used "Synth sounds" in OctaMed or Ageis Sonix. One interesting technique is to use different oscillators with the same waveform and modulation source but with the Oscillators at different octaves.

Phase Shift Modulation

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This changes a waveform by taking a copy, Modifying it and recombining it with the original.

Source selects the modulating wave in the normal way and NEG flips it upside down.

The level determines the level of modulation and can be used to make the effect more or less dramatic.

+ Only/- Only/Both determines how the wave is shifted, + Only means the shift is only forwards - Only makes it backwards, Both means the shift is backwards or forwards depending on the input wave, + and - shifts give different sounds and changing between them gives a very dramatic change.

The Invert, Negative and Reverse options apply to the wave being phase shifted, this is then combined with the original wave giving a wide range of wave shapes, Reverse and Negative do the usual but invert turns the waveform inside out giving a radically distorted wave.

Add/Mul/>/OR determine the way in which the two waves are then combined.

Used with the Negative/Reverse/Invert option and +Only/-Only/Both gives a total number of 128 different types of combination - and thats before the modulation wave changes!

Pulse Width Modulation

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This is an effect found in some analogue Synths but usually only applies to a single wave type. Basically it takes the waveform and splits it in the middle, it then moves the line back and forth causing the two halves to alternately stretch and squeeze. This causes the harmonics found in the waveforms to rise and fall giving a pleasing thickening sound. The level in this case determines the maximum stretch/squeeze, the result here depends on the wave, the more harmonics this has the more pronounced the effect.

Frequency Modulation

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Frequency modulation allows the note to be changed by several octaves creating some interesting sounds, there are some examples of these in the Startup Guide/Tutorial, if you modulate the waveform and frequency at the same time you can get some interesting laser type effects. If you wish to use FM to give a chorusing effect you should set the level slider at 5 or lower - anything higher will give fairly mad frequency sweeps. Please note that this has nothing to do with FM synthesis as found in the Yamaha DX7.

+ Fix & - Fix give you a couple of additional options which are designed to sound something like Oscillator Sync on analogue synths, Fix prevents the pitch of the waveform changing but allows the waveform to change, this gives a harsh sound a bit like Pulse Width Modulation.

## 1.14 Envelope Generators

ENVELOPE GENERATORS

These actually now work. They produce an envelope determined by the sliders on the right and it's length is as long as the sample itself. The interesting bit is when you modulate it.

Making an envelope

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The principle here is the same as Ai but should be somewhat less con fusing. An envelope is made up of points which slope into one another. Setting the Envelope point sizes is done by changing the depth sliders. The lengths between the points is set by the length sliders, these work by ratios so setting them all to 1 gives the same result as setting them all to 256. If you don't want a slope between points set the length to zero, this makes the value jump.

The first point is set at zero so if you want to start at a different value set the first length at zero. The Envelope Default is like a standard Attack/Decay/Sustain/Release type envelope but much more complex types can be generated.

Envelope Triggering & Modulation

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This allows the Envelope to be started at different points or even used as a waveform. There are 3 options:

First Trigger only allows the Envelope to be generated only once but the input determines where this is. It is triggered by a crossing of zero in a waveform.

All Triggers allows the envelope to be started multiple times however the envelope length remains the same.

Modulation allows the envelope to be used in a completely different manner. In this case the envelope acts as a waveform, the input waveform determines which part of part of the envelope is output, changing the shape of the input wave changes the shape of the enve lope wave, the envelope wave also takes on any changes made to the input wave so if it has been frequency modulated the envelope also becomes frequency modulated.

Amplitude Modulation

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Normally a Envelope is used to modulate the level of something such as a volume or whatever but in this case amplitude modulation is directly applied to the input wave.

The input wave can be used entirely or half only:

Full wave uses the entire input wave for modulation.

Half wave only uses the top half of the input wave.

There are two options for modulation:

Multiply multiplies the envelope with the input wave.

Add adds the two together.

## 1.15 Mixers

MIXERS

There are 6 mixers for processing and a further 2 for Left and Right output, They can all be used within the program as sources and all have amplitude modulation.

At the top are 6 input sources along with sliders for setting their levels, To retain volume levels the inputs levels are calculated as a ratio, what this means is that setting the inputs all to 1 is the same as setting them all to 256, the sliders allow you to set the relative levels of inputs not the total level.

The Mixers can also be Amplitude modulated, the level slider here determines the level of modulation and the NEG option flips the modulating wave.

## 1.16 Modifiers

MODIFIERS

This is the first Module that does some real processing.

The input is selected at the top along with some basic wave changing options

Full Wave makes negative parts of the wave positive.

Half Wave Zeros the negative parts of the wave.

Invert turns the entire wave inside out.

Distortion Modulation

---------------------

This allows you to distort a sound but also to modulate the level of distortion. In addition to this there is a contortion option which gives a different form of distortion which can change a perfectly normal waveform into pure noise.

NOTE : The distortion level is set by the level slider, this applies even if there is no input, setting it to the default of 1 has no effect on the sound, the slider is in binary values so that 2 doubles, 3 quadruples etc..

Wave Modulation

---------------

This allows the wave shape to be modulated, depending on the two waves used the result can be subtle or dramatic.

Greater gives the greater of the two waves.

Lesser gives the smaller of the two waves.
Stutter uses the modulating wave to determine which parts of the main wave is zeroed, if the modulating wave is zero the main wave becomes zero.
Rectify uses the modulating wave to change the sign of the main wave. If the modulating wave is - the main wave is made - and if + it is forced +.

## 1.17 Wave Shapers

WAVE SHAPER
This is another module which changes the shape of the waveform.
The first stage is the Phaser, this is similar to a Phase shift but it changes over time, The waveform is Phase shifted by a value which is calculated by the program, The Phaser slider determines the maximum change but this moves between it's value and zero, the bigger the value the longer this takes, this gives everything from a strange wave shaping effect to a chorus type of sound.
The Level determines the amount which is added back.
The Phase/Both determines which of the waves are made negative, whether it is the phase waveform or both.
To switch off the Phasing set the Phaser level to zero.
Limit Modulation
----------------
This limits the change between individual samples, this shapes the waveform into a triangular type wave, the limit determines the amount of limitation, the lower the value the smaller the limit and the bigger the change. The limit can also be modulated by an input giving yet another interesting sound.
Standard - Normal limitation.
Add - Adds limited signal to original.
Multiply - Multiplys original and limited signal.
Shape Quality Modulation
------------------------
This changes the digital quality of the waveform, doing this adds harmonics to the wave form and can be used to good effect.
There are 4 options here:
Bandwidth - Modulates the bandwidth of the samples, this gives an effective sample rate output between 22 and 344Hz, sweeping this

makes a very interesting sound, at lower sample rates additional harmonics are generated and these are swept with the length of the samples. The output is very dependant on the sample being processed.
Bits - In Ai you can set the number of bits in a sample from 16 to 1. The less bits the more noisier it becomes. Here is works by division so the change between is less sudden, it's more of a gradual change, this however can also be swept giving an interesting sound but completely different from bandwidth changes.
Both - This is both Bandwidth and Bits modulation together.
Both Opp - This is both Bandwidth and Bits again but this time as one goes up the other goes down producing a very different result from the above.

## 1.18 Filters

FILTERS
The filter consists of a 7 band Graphic Equalizer, this splits a sound into 7 frequency ranges and allows you to set them, the ranges go from very low frequency bass sounds up to very high frequency sounds.
The level slider sets the output volume, the EQ can sometimes give very low volume results so the slider can be set up to 512 to increase the output volume.
Depth Modulation
----------------
The Depth Modulation mixes the input and outputs from the filter changing it's effect on the sound, this modulates between full EQ and none the volume can also change quite a bit what modulating.

## 1.19 Amplifiers

AMPLIFIERS
The Amplifiers allow levels to be set with the Level slider.
Amplitude Modulation
--------------------
The Amplitude can be modulated with the second input and the level set with the slider.
Wave Conbination
----------------

Wave Combination allows the main wave to be combined with the
third input in a number of ways producing some very different
sounds.
Add simply adds the sounds
Subtract gives the opposite of Add.
Multiply makes the wave change over time.
Divide gives a very thin sound.
AND gives a thick sound.
OR gives very similar results to AND but slightly different.
XOR gives a different type of thick sound.

## 1.20 Delay

DELAY
This gives a sweeping effect, it is limited to a short delay which
is set in the Level slider.
Add/Sub selects the way the delay is recombined.
Delay Modulation
----------------
The Delay can be modulated with the Delay Modulation input, allowing
a sweeping effect to be generated. Although if you feed a frequency
modulated sound into the delay with no modulation it also gives a
sweeping effect.
Depth Modulation
----------------
The Depth Modulation allows the level of the delay to be modulated
with another waveform, this mixes the effect with the original signal.
Phaser
------
The Phaser is much the same as the Phaser in the shaper. As with
the shaper a smaller value gives a faster effect.

## 1.21 The Tool Box

TOOL BOX
The tool box is the module where what was left went, It includes
five separate effects.
Noise Generator
If the level is set above 0 the incoming wave is byte reversed giving a

very noisy output, feed in a low frequency waveform and you will get an interesting sweeping effect, mix two together and feed them in and it's even better.

Level sets the level of the output, 0 has no effect on the waveform.

Sample and Hold

This is an effect found on old Synths which is very recognisable but not by listening to the output. Feed the output into the frequency modulation section of an oscillator and you get a 70's computer blipping away.

The hold length can be very long so a bit of messing around with the slider may be required.

The Input determines the length that the hold is held for. Basic Synth 12 and 13 show this in action. 0 disables this effect.

Amplitude Modulation

This was added here to spice up the sample and hold a little bit, the hold output is set values but the Amplitude modulation allows these values to be modulated. If this is then feed into a frequency modulation input the result can be very weird indeed. You can of course feed this into all the oscillators inputs. Basic Synth 14 shows the effect of adding this after the sample and hold.

Basic filter

The next two sections of the tool box combine to make a tunable resonant filter, If you are expecting all the gurgles and blips of a proper analogue filter it's probably better to get the real thing as this isn't the same at all, (although you can program gurgles if you want) the output however is quite interesting as it sounds like no other filter.

The filter frequency is modulated by the modulation input and the level is set by the level slider. A number of filter modes are available.

Setting the level to 0 disables the filter but if an input is still present it is still used to determine the resonance frequency.

Low Pass - Keeps only low frequencies.

High Pass - Keeps only high frequencies.

Band Pass - Keeps only mid band frequencies.

Band Cut - Cuts the mid band.

Notch Pass - Keeps only a small number of midband frequencies.

Notch Cut - Cuts only a small number of midband frequencies.

Frequency - Sets the frequency at which the filter operates, if there is an input both of these determine the frequency, by setting the frequency to zero only the input determines the filter frequency.

Level - Sets the depth of the filtering.

Resonance
The Resonance is controlled separately from the filter, the depth is set by the slider but the frequency is controlled by the filter modulation input/slider. The modulation input to the resonance section modulates the depth of the resonance, it is perfectly feasible to have a filter with resonance fading in and out and it is also possible to have resonance without the filter, something which isn't possible with an analogue filter, in this mode the resonance will give similar results to the delay.
setting the level to zero disables the resonance.
Level - Sets the level of resonance.
Delay/Filter/Both - Determines the type of resonance used.
Low Q/High Q - Determines the sharpness of the filter used for resonance in the Filter and Both options above.
The low pass filter with resonance can be a touch crackly but this can probably be used to effect.

## 1.22 Left and Right Mixers

LEFT & RIGHT OUT
Both of these work but the outputs are combined and give a mono output. They work in the same way as mixers but unlike the other mixers you do not need to take an output from them as this is automatic.
It is however possible to use the output as a modulation or sound source in the program.
Stereo Output
When the Program creates a stereo sample the Left and Right outputs are used to create each stereo side, in all other modes the outputs are added

## 1.23 Problems

Q & A
Q1 - How do I stop a sample from playing?
A - You don't! The entire sample will play then stop.
Q2 - Why are clicks or glitches at the beginning or end of the sample?
A - This is due to the way the program but it can be reduced by use of the Envelopes by setting the sound to begin and end inside the area of the sample. Alternatively a Sample editor can be used to cut or zero the offending parts of the sample.

Q3 - If I save a sample and use it in a music program it sounds far to slow and low pitched.

A - Synthetica saves it's samples at 44.1KHz and this can be too high for some sample programs, the way to get round this is to resample to a lower sampling rate or retune the sample. If the program has a Transpose function (like OctaMed) this can be used.

If you have Aural illusion there is a manipulation called "Octave -" which will cut the sample rate down to 22KHz where it can then be transposed, this is present in v1.1 of Aural illusion which is now available as freeware.

Q3 - Why does sound playback stutter?

A - Playing back 16 bit sound with 8 bit hardware requires it to be converted in real time, without using "clever" playback routines this becomes quite a chore for the CPU and anyone with a 68000 may have stutter. This will probably happen if you are running other programs so the way to avoid it is to stop the other programs.

Q4 - Why wont the program play a sound?

A - If the sound channels are in use the program will run without them, When this happen the Play button is ghosted.

Q5 - I have found that when I press play the program can freeze, why?

A - This has happened in some beta versions but the bug was fixed, it may still occur but shouldn't, if it does - or if you have any other serious problem please let me know.

Other Problems

If you find any problems with this program please let me know of them so I can get them sorted, if you do find a bug please tell me what Amiga / configuration you have and if possible when/how the bug occurred.

Updates

No further updates are planned in the near future however a version 2 is planned as is a real time version. A PC port is also likeley.

Suggestions

If there is anything you would like to see added to the program please let me know of these, and I'll see if I can add them.

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## 1.24 Changes in v1.1

Changes to Aural Synthetica v1.1

This section is for the benifit of those upgrading from v1.0, most is
repeated elsewhere but this section is to stop you having to search for
the changes through the manual.

v1.1 of Aural Synthetica includes quite a few additions, changes and the
odd bug fix, heres the main changes:

Virtual Memory support

----------------------

The first change you will notice is an extra button on the memory selection
screen. This allows you to switch between Normal and Virtual memory,for
this to work you will need to be running a virtual memory program such as
GigaMem or VMM, your CPU will require a MMU for such a program to work.
If you have such a system AS will detect the difference between normal and
virtual memory and allow you to choose which you wish to use, you should be
aware that virtual memory is considerably slower than normal RAM.

Pull Down Menus

Save Patch + Waves...

---------------------

You may notice that if your patches use waves which you created they don't
sound the same if saved then reloaded, this is because v1.0 can't actually
save the waves it creates. The new selection allows you to save the patch
and any waves used in it, these files can be considerably bigger as the
waves take up 2K each.

Audio Device...

---------------

This allows you select between the Amigas output, no output and any other
devices supported, at time of writing no other device is supported, Aura
support was less than successful but may make it into a later version...

Module Changes

Oscillators

-----------

One change has been made to the Oscillators in the Frequency Modulation section. A new type of Modulation called Fixed Pitch Modulation has been added, this is designed to give similar-ish results to an effect on analogue Synths known as Oscillator Sync. The waveform changes the pitch does not, this gives a harsh sweeping sound not entirely unlike Pulse Width Modulation. The settings allow the wave to get bigger or smaller but not both, normal Frequency Modulation is disabled while either of these settings are used, the default is off.

Shapers

-------

The middle section of the shaper allows a feature called Limiting to be used which only allows the waveform to change by the amount set (or modulated). In v1.1 the options Add and Multiply have been added, these take the limited result and either add it to or multiply it by the original sound, both can create slightly harsher sounds although this depends on the input wave. The default is standard.

Tool Box Filters

-----------------

The biggest difference by far in v1.1 is in the filter & Resonance sections of the Tool Box. The filter itself has 2 extra modes, Notch pass and Notch cut which only pass or cut a very narrow band of frequencies. In addition to this thee is also a Frequency slider which allows you to manually change the frequency at which the filter operates, previously this could only be done with the modulation input.

Resonance

---------

The resonance section now contains an extra two sections, the first selects the type of resonance in use, the default is Delay which was used in v1.0 but there is also options for Filter and Both. If you wish to recreate an analogue sounding filter these will give a more authentic sound.

When the Filter or Both options are used the second selector is used to determine the type of filter used creating the resonance effect. This gives options of low Q or high Q.

The High Q option can go into heavy distortion depending on the input, this in itself can be useful at depending on what you wish to do.

Rendering Changes

Quality

-------

AS uses a very complex retuning algorithm for keeping notes as accurate as

possible without distorting the waveform, however it is now also possible
to use a low quality algorithm which can cause the waveform to become
distorted at times, this gives a less pure sound. Try Basic Synth 25 with
both options and you will hear quite a difference. Another thing to try
is to sweep the frequency of an oscillator, the Less setting will give a
staggered sound as the waveshape distorts at certain frequencies.

Length

------

The length of the sample generated in v1.0 was determined by the amount of
memory selected upon loading the program. This slider now allows to create
a sample of any length between 1 and 100% of allocated memory. for testing
different options is is now possible to create short samples quickly then
create the full sample later.

Instability

-----------

Analogue synthesizers are famed for their inability to stay in tune for any
reasonable length of time, this in itself was part of their sound and this
slider has been added to recreate that effect, a small random retuning is
added to the waveforms when the oscillator creates them, depending on the
slider setting this effect can anything from barely audible to quite
"ill" sounding.

Rendering speed

---------------

Unfortunately it has not been possible to create a massive increase in
rendering speed however as you may notice the program is now faster. The
length option will probably give the biggest difference in that it allows
short test samples to be generated.

Glitches

--------

There tend to be clicks and the like at the beginning of samples created
in this program and this has now been reduced considerably, unfortunateley
it has proved impossible to remove the problem completeley.
Some programmers do listen to reviews you know!

## 1.25 Miscellaneous

Aural Synthetica v1.1 was written entirely in C by Nicholas Blachford using:
Amiga A1200 68030 50MHz 10MB +60MB HD +Zip 100MB
SAS/C v6.2 (v1.0 - 31,508 lines, 814K) (v1.1 - 32,680 lines, 843K)

GUI code generated by GadToolsBox v2.0

(no it didn't like the Patch Programmer either)

Many thanks to :

SAS Institute for SAS/C.

Jan van den Baard for GadToolsBox.

The old Commodore for Enforcer and other tools.

Sound on Sound and Future Music for various ideas and articles

on Modular Synthesizers.

All at MidiCraft and MUG.

Amiga Technologies for finally showing they're doing something.

The Beta Testers:

Kevan Craft

Gareth Craft

Dave Sullivan

Rob Baxter

Fergus Duniho

And special thanks to :

Julian & Sue Sula for selling the program for me, a few trillion ideas and

putting up with me on the phone!

Jimmy Hill for selling me his Monitor (EURO 72 is much better on the eyes).

Amiga Shopper for putting Aural illusion v1.1, Aural Synthetica Demo &

Patch disc vol1 on the cover.



This document by Nicholas Blachford 16/4/96.

## 1.26 Aural illusion v2.0

You've made your new sounds but now what...

Edit, Create and Mutate your samples in :

Aural illusion v2.0

Now with 30 Variable Effects including:

--------------------------------------

Time Stretch,

Graphic Equalizer,

Mixer,

Resonant Filter,

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Gate,

Reverb,

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Chorus,

Flange,

Phase Shift,

Stereo Processor,

and more...

55 Manipulations including:

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8 Tunings,

6 Modulations,

8 Filters,

12 Arithmetic/Logic functions,

9 Waveshaping functions,

and others as well.

16 Bit "Morphing Wave" synthesizer with:

---------------------------------------

8 mixable wave types,

4 step Wave Morphing,

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Envelope Generator,

Phase controls for improved sound.

Editing Functions including:

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and many more.

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RAW 8,

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All processing done in 32 bit format,

samples held in 16 bit format.

Requires Workbench v2.04 or greater and 2 Meg RAM.

All this for a mere £21 inc P&P (UK Pounds).

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## 1.27 Amiga Music Disc Magazines

Are you an Amiga MIDI musician or perhaps an OctaMed dabbler?

If so there are two disc magazines which you may be interested in:

Total Irrelevance (TI) by the OctaMed Users Group

-------------------------------------------------

TI is mainly based around OctaMed users but caters for others programs as well.

You get a selection of Articles, News, Tutorials, Reviews and usually a couple of Octamed Modules to play. There are reviews every month of PD Mods and Members Mods so you can get your Mods judged.

There are also special offers and a MUG club you can join with even more offers. In addition to this is the OctaMed BBS where you can up/down load Mods.

MidiCraft Magazine

------------------

MidiCraft is a more general Amiga music magazine covering just about everything music related, the mag is mostly based around articles but also reviews users and commercial Tapes and CDs. As with TI you will also find Mods but you will also find Music X and GM files for playback from external Synths.

MidiCraft also does a Sample disc to accompany each magazine disc.

Both Magazines are available from:

Seasoft Computing

Unit 3,

Minster Court,

Courtwick Lane,

Littlehampton,

West Sussex,

BN17 7RN