

AudioFAQ

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Chapter 1

AudioFAQ

1.1 Audio FAQ -- A guide to sound effects

Audio.FAQ -- A guide to sound effects (v1.0)

- (c) 1996-7 by Jarkko Valtjus-Anttila

1. Information

- 1.1 Disclaimer Read this before anything!
- 1.2 Introduction And this too.
- 1.3 Author Author information.

2. Effects (in alphabetical order):

- 2.01 Backwards 2.09 Minimize
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- 3.1 Box 3.4 Triangular
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4. Miscellaneous

- 4.1 Drawing How to draw samples.
- 4.2 SinED v1.0 Introduction to SinED v1.0.
- 4.3 Pictures Some audio related pictures.
- 4.4 Interpolation How to interpolate.

1.2 Disclaimer

I cannot take any responsibility about misspellings or other flaws that may cause trouble if you rely on the information described in this FAQ file. I've spend hours and hours doing a sound effect

generator so all of the ideas in this guide are tested in real-life and notified to be working as they should be.

I cannot either say that this information about the sound effects is absolutely correct. This is because I've not found any info myself and I have had to improvise. So, you could say that in this FAQ file I've gathered my own ideas and solutions for problems in the field of sound engineering.

There might be easier solutions for the effect calculations described below, but again that's not fault. Don't read this FAQ like a bible, but still use it as your information resource.

If you have audio related problems, I'm very willingly to help. Just contact me and describe the problem.

Note that all effects described in this FAQ file are used by my program SinED . It can be found from aminet from dir mus/edit/SinED.lha. When reading this file, I'd suggest that you downloaded the editor and tested the effects with it.

1.3 Introduction

Introduction

Well, this whole thing started a long time ago, when I was in an urgent need of a capable sample-editor. Unfortunately there was not a single editor available that would meet all of my needs. I was so desperate that I even pirated the AudioMaster but only to see that it was worse than most of the others. At that moment I decided to write my own editor that would do everything I want. At least this time I'd get what I want. (Besides my C skills needed a practice :D)

Before that I didn't know anything about processing the audio data, I began my research, but how odd: it seemed that nobody knew anything about the more complex effects. Hell, everyone knows how to flip the data backwards, but when it comes to for example modulation routines, I didn't get any info. That's why every single routine described in this guide is based on my thoughts and ideas that have come to my mind during the programming sessions. I cannot guarantee that these instructions are the best around, or that they were even correct, but they give a correct result and that's enough for me.

That's how this project started. If you own an Amiga with KS2.04+, then I highly recommend you to get my sample editor from aminet. At this moment (16.1.1997) the editor is not uploaded there, but it will be soon and the name will be:

mus/edit/SinED.lha v1.0 The best sample editor/generator so far.

With the program you can explore the routines and listen how they hear like. If you can't find the program from the aminet, you can always request it from me . I'd be more than happy to send it to you.

You're possibly wondering why I've not made myself clear with example routines for example with C language. This is because the sources of SinED are already included in it's package. Download it and take a look at them.

Have fun with this guide. I'd be more than happy to hear from your thoughts about this file, and if you knew new routines to include in this guide, then please Contact me!

Signed: Jarkko Vatjus-Anttila

1.4 Author information

Author info:

If you have any problems with this file or audio problems that you need help with, then don't hesitate to contact me.

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Note: My EMail account will expire on the 1st of June 1997. Before that you can contact me, but after it, it might be more difficult. My new account will open on the 1st of September 1997 but the exact address is still unknown. However, if you really need to contact me after June, then drop a note in some of the comp.sys.amiga.* newsgroups. I'll most probably notice it there.

1.5 Backwards

Backwards

I think this one needs no other introduction than the name. The sample data is simply flipped backwards. The last value is switched with the first one, the second last is switched with the second, and so on until every sample value is switched with its corresponding value.

1.6 Blur

Blur

The blur effect mixes every sample value about 1%-15% from the sample's amplitude. If you are processing 16bit sample data, the length of the amplitude would be 65536 and the maximum blur value $0.15 * 65535 = 9800$. Now, create a random number between 0 and 9800 and then randomly add or subtract it from the current sample value. Loop this procedure until you have affected all the sample values, and by result you have a blurred sample. This gives a foggy feeling to the waveform.

1.7 Boost

Boost

Boosting means that the sample peaks are increased and sharpened. One of the methods to create an effect like this, is to use a linear interpolation. That method applies to boost effect more than better because we actually are calculating first degree polynomials. One thing that remains, no matter how you boost is that the first and the last value in the sample are not affected in anyway.

Create a interpolation line from the second sample value to the fourth one. That's right, skip the third one. Then calculate the value for the third samplevalue with the interpolation line. This would now be the value that the third sample value would have if the three values were in a row. However, it's highly unlikely that the three values were in a row, so now you have to calculate the distance of the third value from the interpolation line. This can be easily done by subtracting the interpolated value from the real value.

Now we have the distance stored somewhere for later calculation. Because we had to increase the sample peaks in the boost operation, we need to increase the distance for the third value from the interpolation line. In this way the peak rises and sharpens. By this point you can add a boost-factor in the game. Use the factor to add a multiply of the distance to the third sample value. For example if the factor was 2 (which is very high) the new value for the third sample value would be:

$$3rd\ samplevalue = 3rd\ samplevalue + factor * distance$$

Usually the factor is a fraction.

You have to take special care when boosting the sample, that the sample values may never increase or decrease so much that they would change their sign. For example if the 16bit signed sample had a value of 15000 before the boost operation, then the maximum positive boost would be $32767 - 15000 = 17767$ and the maximum negative boost value would be $-32768 - 15000 = -47768$. Graphically thinking this means that the sample peaks in the picture of sample wave may

never raise over 32767 or lower below 32768.

After calculating the value for this sample value, go to calculate the next one. Jump one value forwards and repeat the calculations. Note that you jump only one, not three values forward. This is because when calculating this effect, all of the values in the sample have to be affected during the boost, but from the bunch of the three values only the middle one is affected. That's why you jump only one value forwards, so that the middle value is now the fifth one. Loop to the end of the sample.

Note too that you need two buffers for this operation. This is because if you calculated values from one buffer to the same, over writing the values, only the first calculation would be accurate, the rest would be something else. This is because the previous calculation would always affect the next one. This may no happen, so two buffers are required.

Figure 1: White noise

Figure 3: Boosted white noise

1.8 Echo

Echo

Echo sounds cool, but is nothing more than mixing waveforms together. With echo you need 3 values: delay, decay and repeat. Delay is the time period how long is the difference between echos. You may decide yourself if this value was seconds of sample values or something else. Decay is a value that affects the echo volume. Usually this is percentages. For example if the value was 80% it would mean that every time the echo is repeated, the volume would be only 80% from the previous echo volume. Repeat tells only how many times the echo is repeated.

Example: Delay = 300 samplevalues, decay = 80, repeat = 5

Delay value is this time samplevalues, and it means that first echo is repeated when the sample has been played for 300 values, the second echo would happen after 600 sample values and so on. Take the original waveform to another buffer and decrease it's volume to 80% from the original. Then mix it to the original waveform starting from the samplevalue 300. For the second echo dercrease the volume from the previous echo again to 80% and mix in to the original sample from the sample value 600. Repeat these mixing procedures as many times the repeat count says and you most probably have a fine echo. ;)

1.9 Filter

Filter

Filter is just the opposite to boost . Calculate the values with interpolation line, but instead raising and sharpening the sample peak, lower and smooth it. This makes the sample to go smoother.

I think I don't need to explain this further. Take a look at the boost operation.

Figure 1: White noise

Figure 2: Filtered white noise

1.10 Flip

Flip

Flipping is flipping the samplewave upside down. :) If you are processing 16bit sampledata, topmost value in the sample window is 32767. Every sample value (unsigned 0 -> 65535) must be thrown to other side of this value. This can easily done for example with this algorithm:

```
newsamplevalue = 65535 - samplevalue
```

or for 8bit sample data:

```
newsamplevalue = 255 - samplevalue
```

1.11 Maximize

Maximize

In maximize routine the main purpose is to raise the volume of the waveform without trashing the sample. This is done by calculating the smallest value that every samplevalue can be multiplied with, without the sample value changes it's sign. If the original value was positive, then find the value that multiplied with the original value gives a result 32767. If the sample value was negative, then Find a value that multiplied with the original value gives a result -32768. Loop the whole sample and take the *smallest* number to memory. If this value is 1 then the sample is already at its maximum but if the value is something else (usually a floating point value between 1 and 2) then multiply every value in the sample with this value. Resulting you have a maximized sample.

1.12 Minimize

Minimize -----

Minimize is the oppoiste to maximize . This time the main idea is to lower the volume without losing any of the sample peaks. This can be done by doing following calculations: If the original samplevalue is positive or zero, then calculate a number that would have to be substracted to get a zero. If the sample value was negative, then calculate the value that would have to added to get a zero. Take the smallest number in memory and then loop the whole sample with adding the value to negative samplevalues and substracting the value from the positive sample values. Resulting you have a waveform with minimiun possible volume.

1.13 Mix

Mix ---

Mixing is copying two samples together. Let's assume that you have two signed waveforms (they have to be signed!) You might have heard about the interference between two waves. This works exactly the same. Add the samplevalues together and divide the result with 2 to get the sample approximately to the same level than the two sample before mixing. This is not enough though. The volume goes too low, so you have to add a factor to both samples to the mixing procedure. If the volume is wanted to get to the same level than the previous waveforms, the factor should be about $\sqrt{3}$. With this information the mixing routine get this form:

```
newsamplevalue = (1st_samplevalue*1.7 + 2nd_samplevalue*1.7) / 2
```

Basically thinking the mixing is to calculate the average for two or more samples.

1.14 Modulate

Modulate -----

Modulation is one of the more exotic routines and you can get really weird sounds with this one. The idea is first to create a base waveform. Usually this is a sine or similar. The length of the base wave is almost obsolete, and I use a wave with length about 2048 bytes. After creating the base wave, get the skip value from somewhere. This value tells you how many samplevalues you have to skip in the base sampledata when you proceed to the next sample value in the original sampledata. Of course, when you reach the end of the base sampledata, loop back to start. Everyone from the original sample values have to be processed.

The modulation is done by multiplying the current samplevalue with the current base sample value and then dividing with the half of the sample amplitude. That depends on the sample accuracy you are using. If you use 8bit data, then this value would be 128 but for 16bit data this value is 32767.

The algorithm used to modulate is:

```
newsamplevalue = base_samplevalue * samplevalue / 32767
```

The datas have to be signed in this case too, and the accuracy of the samples has to be the same.

1.15 Shadow

Shadow

I'm not quite sure if this one is obsolete or not. Well, the idea is to reset every other sample value to zero. The sound changes to hear like insects in some cases. Perhaps this one is my weird imagination. :D

1.16 Shift up

Shift up

Shift up and Shift down exists because of some dumb samplers that sample the data in a way that the result is no longer steadily around the base level. With these effects you can adjust the sample to the right position. This does not affect the sound at all, if sample peaks are not lost.

Shift up and Shift down can be linked together to a routine that would automagically calculate the best location for the sample waveform. If this kind of effects exists in the program it's usually called "Normalize" or "Normal DC"

1.17 Shift down

Shift down

Exactly the same as the Shift up but this one shifts the data downwards.

1.18 Volume slide

Volume slide

Volume slide is for changing the sample volume smoothly from some point to another. This can be done for example by asking the user for percentages that the sample volume should be in the beginning from the original sample and in the end. Then create a factor that is changed all the time sliding forward on the sample. For example to slide a volume from 200% to 50% the factor would get values from 2 to 0.5. Just calculate the correct factor value for each and every sample value and multiply every value in the sample with it.

Always remember that if the factor is greater than one, the sample can be clipped and distorted.

Figure 4: An example of a small talk sample that has been volume slided from 300% to 0%. You can see the distortion in the beginning and the fading in the end.

1.19 Downsample

DownSample

Downsample means that the sample is changed in a way that it sounds lower although it is played with the same frequency. As you might imagine this is done enlarging the sample buffer. If you have a sample that you would like to hear twice as low without lowering the playing frequency, you would double the sample buffer and copy the sample values from the source sample into every other spot in the target buffer. Then fill the empty spots in the target buffer for example with interpolation lines. Just calculate one for every single space the target buffer contains and fill them. In this way the sample is longer and that's why it sounds lower although you play it with the normal frequency.

If you don't want to halve the sound frequency, you can always allocate for example a buffer that is one third bigger than the source one, and then interpolate every third value. This would enlarge the buffer size one third larger and then the frequency would drop one third. Simple as that.

1.20 Upsample

Upsample

Upsample works just like downsample, except that this time the buffer size is reduced instead of enlarging. This is more risky than downsampling because when the buffer size is reduced, some of the

sample information must be thrown away. Too much upsampling and the sample is distorted.

Halve the buffer size, and the sample hears like twice as high. Reduce one thirds and the frequency raises one thirds, etc.

1.21 Box wave

Box

This one is simple. Just fill the half of the buffer with the value that is maximum positive value for your current sample accuracy. For 8bit samples this would be 127 and for 16bit samples 32767. Then fill the latter half from the buffer with the minimum value. For 8bit samples this would be -128 and for 16bit samples -32768.

Figure 6: Example of a box-wave.

1.22 Saw wave

Saw

This one is as easy as the box wave if you consider the sample buffer to be unsigned. The scale would then be for 8bit samples from 0 to 255 and for 16bit sample from 0 to 65535. Then just create an interpolation line of which starting point is (0,0) and ending point is (sampleend,65535). Sampleend is the sample length in sample values. After creating the interpolation line just calculate with it the values for every space in the sample buffer.

Figure 7: Example of a saw-wave.

1.23 Sine wave

Sine

The period of the sine is 2π . Divide this value with the length of your buffer in sample values. This gives you the delta angle that you must jump forwards when calculating the wave. Begin from 0 and end to 2π .

Sample code:

```
deltavalue = 2*PI/samplelength;
for (loop=0; loop<2*PI; loop++)
    buffer[loop] = sin(loop*deltavalue);
```

Figure 8: Example of a sine-wave.

1.24 Triangular wave

Triangular

Triangular wave is similar to saw, but this time the interpolation is divided into 3 sections. You have to calculate three lines, with following coordinates:

	Start	End
	-----	---
Line 1:	(0,0)	(samplelength/4,32767)
Line 2:	(samplelength/4,32767)	(3*samplelength/4,-32768)
Line 3:	(3*samplelength/4,-32768)	(0,0)

Then fill the first part with the first interpolation line, the second part with the second line and the third part with the third line. As you can see, the first line raises from the base position to the maximum, the second lowers from the maximum to the minimum and the third one raises from the minimum to the base position. When these three lines are connected, there appears a triangular wave.

Triangular wave is similar to sine wave.

Figure 9: Example of a triangular-wave.

1.25 White noise

White noise

White noise is just random numbers with the complete scale of your current sample accuracy. Just write a decent randomnumber generator and you'll create a white noise just fine.

Figure 1: White noise

1.26 How to draw a sample

How to draw a sample

Pictures Some waveform pictures here!

This can be done easily if the sampledata is in signed form. Imagine a box of this kind:

32767 +-----+



Where the *** line is the midline. Each horizontal line in that box represent its own sample value and the *** line is the zero. Consider the box as coordinate axles where the *** line is x-axle, the leftmost vertical line is the y-axle and the X is the origin. The length of the x-axle is the same as the length of the sample and the length of the Y axle is the same as the amplitude of the sample. For 8bit samples this would be 256 (-127 -> 128) and for 16bit samples 65536 (-32767 -> 32768). Of course these kinds of sample boxes cannot be drawn, so the you have to scale one to fit on your screen or window.

Ok, the drawing then is simple. Just take the value from the sample and move your pen to that spot on the sample box. If the first value is (8bit sample) 69 then the coordinates are (0,69). If you imagined the sample box to be the conjunction of the coordinate axles the only think you need to do, is to scale the coordinatepair (0,69) to your screen coordinates. Check all of the sample values and connect the plots on the screen with lines, and there you have a sample wave drawn on your screen.

Figure 5: Simple samplewave example.

Drawing speedups:

- Calculate first how may sample values you have to skip before drawing. I mean that if you have a screen width 640 pixels and the sample length is 102400 bytes. In this case only every 160th value would have to checked and displayed in the screen. In this way you only need to check 640 values in total from the sample instead of the 102400. The speedup is remarkable.
- You can use just pixels for drawing, too. Lines are not slow when displaying a sample, but pixels are always faster.

Special notes:

- Let's assume that the screen width is 640 pixels. If you are drawing a sample which length is smaller than 640, you have to skip pixels on the screen instead the vaules in the sample, to fit the sample in the 640 pixels. This time you cannot connect the values with lines because it wouldn't be drawn right then. If the two values were just connected with a line, it would tell the user that the sample actually wasn't so short and it contained values between those connected samplevalues, even it didn't contain them. The idea is that when sample value occurs,

it lasts until the next value comes up. That's why you have to draw a horizontal line until the next sample value and then connect the end of the horizontal line to the next sample value with a vertical line, this makes the sample wave to look like a stairs in most cases, but at least it's the truth.

1.27 SinED v1.0

SinED v1.0

This is the .readme file of the SinED v1.0

Description: V1.0 The most powerful sample editor/generator ever made!

Author: Jarkko Vatjus-Anttila <quaid@kempele.fi>

Uploader: Jarkko Vatjus-Anttila <quaid@kempele.fi>

Version: 1.0

Dir: mus/edit

Welcome to SinED v1.0, the most powerful sample editor/generator so far. :)

(c) 1996-7 Jarkko Vatjus-Anttila <quaid@kempele.fi>

This program is without a doubt the most powerful sample editor generator so far. The reason this is written, is because I needed a decent sample editor, but the ones available were more or less pathetic. It doesn't have to be that the only good sample editors are available only for those who have a DSP installed in their machines. SinED will fix that problem.

Just take a look at the list about the features that SinED provides to you. You will not believe your eyes.

Hype? Perhaps, but still take a look. :)

Advantages:

- * It's free.
 - * All calculations are made in 16 bit accuracy. This ensures that all calculated effects and operated samples have the best possible quality. 8bit samples are just an option.
 - * Currently SinED is capable to create Sines, box, triangular and saw -waves, not forgetting the white noise (and clear :).
 - * This version includes following effects: Backwards, Flip, Shift up, Shift down, Shadow, Blur, Boost, Filter, Maximize, Minimize, Mix, Modulate, Tunnel echo, Repeat echo, Convert and Translate.
-

- * Some effects are combined in a way that you can create cool sound sweeps. For example you can combine the filter and the boost options to create for example an opening filter. I remind that all effects are smooth due the 16bit accuracy.
- * Samples can have unlimited length. Only your amount of memory will slow down you work. It works with virtual memory too, although only VMM is tested.
- * Samples can be displayed in four different drawing styles. Lines, Pixels, solid and inverse.
- * Most common sample format are now supported, like IFF-8SVX, IFF-16SV, WAV and AIFF. SinED includes routines for saving and loading mono and stereo modes. Not surround.
- * Includes two stereo channels. Effects can be calculated for both channels or just for other. Channel lengths are independent, so both channels can contain for example their own sample.
- * Multilevel undo option. Only your memory limits the actions.
- * Of course the samples can be played, with user definable settings.
- * Samples can be edited by hands.
- * It's far more better than for example AudioMaster, the most pathetic sample-editor ever made. Take a look at this one!!!!

Download this now! It's worth it.

1.28 Audio pictures:

Audio pictures:

Here are all of the pictures gathered from this guide onto this one page. All pictures are created with SinED v1.0 and grabbed with QuickGrab.

- Figure 1: White noise
 - Figure 2: Filtered white noise
 - Figure 3: Boosted white noise
 - Figure 4: An example of a small talk sample that has been volume slid from 300% to 0%. You can see the distortion in the beginning.
 - Figure 5: Simple samplewave example.
 - Figure 6: Example of a box-wave.
 - Figure 7: Example of a saw-wave.
 - Figure 8: Example of a sine-wave.
 - Figure 9: Example of a triangular-wave.
-

1.29 Linear interpolation

Linear interpolation

In this guide I refer a lot to a simple mathematical solving method called linear interpolation. Let's see how it works.

The main idea is to calculate a formula for a line that would match most of the values we are currently processing. In most cases the more complex formula changes to a first degree approximation formula that can easily be used to assist in work. And more importantly, it's far more faster than calculating with the original values. Take a special care noting that this really is just an approximation. You will not get accurate results.

As you might imagine, to create a first degree formula, you need to know the start and the end coordinate pairs in order to calculate the slope and to generate the formula.

Let's assume that the start point is (x0,y0) and the end point is (x1,y1). Then the slope would be:

$$k = \frac{y1 - y0}{x1 - x0}$$

The first degree formula can be written like this, where the plain Y and X are variables:

$$Y - y0 = k * (X - x0)$$

Adding the slope to the function and moving the y0 to right side we get:

$$Y = \frac{y1 - y0}{x1 - x0} * X - \frac{y1 - y0}{x1 - x0} * x0 + y0$$

Everyone can optimize that to suit oneself's needs.

Now when passing values for X, which usually means the sample buffer spaces, you can calculate the value for Y which now represents the sample value. This is the function that can be user for many simple operations.