

Perfect Sound

Stereo Sound Digitizer for the Amiga



SunRize Industries

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Perfect Sound for the Amiga

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Files on Your Disk

Important: Before using any of the programs on the disk you received, make a backup and use the copy, not the original.

PSOUND

The Sound Editor Program

FORMAT__IFF

Text file with technical specs for the IFF format

SOURCE

A directory containing the source code for *Perfect Sound*

SOUNDS

A directory that contains digitized sounds in IFF format

The disk you received does not contain Workbench. If you have a one drive system, you may wish to copy *Perfect Sound* on to a Workbench disk. This is not absolutely necessary since *Perfect Sound* can be loaded by booting with a Workbench disk as usual, then removing Workbench and inserting the the *Perfect Sound* disk. The *Perfect Sound* disk icon will appear, and you can double click it as usual.

If you wish to copy *Perfect Sound* on to a Workbench disk, do the following:

Boot your system as normal (insert Kickstart then Workbench). Double click the Workbench icon to open the Workbench window. Now, remove the Workbench disk, and insert the *Perfect Sound* disk. Double click the *Perfect Sound* icon to open its window. Once both the *Perfect Sound* and Workbench windows are open, simply drag the recorder icon across to the Workbench window.

This method allows you to copy the *Perfect Sound* executable program only, and does not copy the library of sounds.

Welcome

Thank you for choosing the SunRize *Perfect Sound* digitizer. If you have any questions, feel free to call (409) 846-1311 for technical support. Our normal hours are from 9 a.m. to 6 p.m. Central time. For technical support, you can also send mail on People Link to "SUNRIZE."

Using our sound digitizer adds a new dimension to your Amiga. Sound can be added to Deluxe Video or your own programs. If you have a music editor that uses IFF files, such as Instant Music, you can create your own instruments. *Perfect Sound* comes with a complete editor that allows you to modify sounds you record. You can delete parts of a sound, alter the order of a sound, change playback speeds, and even see graphs of a sound.

Please do not forget to send in your registration—*Perfect Sound* is constantly being upgraded.

What is digitized sound?

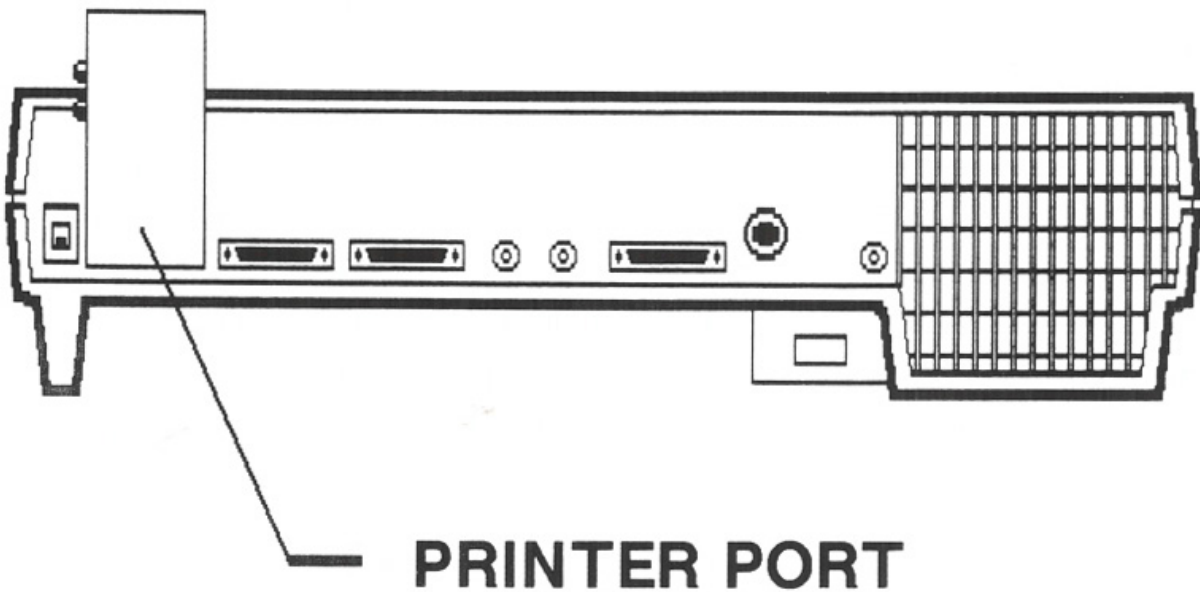
The *Perfect Sound* audio digitizer converts an audio input into a series of numbers that represent the input sound. The input waveform is sampled 5,000 to 25,000 times a second. Each sample is a number from 0 to 255 that indicates the amplitude of the input wave at the instant in time when the sample was taken. The higher the sampling rate you use, the better the sound will be represented — however, higher sampling rates use more memory. Digitized sound uses a lot of memory. For example, two seconds of reasonable stereo sound will easily take 50,000 bytes.

For a more complete discussion on digitized sound, see the Amiga hardware reference manual.

Installation

The *Perfect Sound* digitizer plugs into the printer/parallel port in the back of your Amiga. The two jacks on the side of the digitizer are for connecting an audio source. The bottom one is for the left channel, and the top one is for the right channel. They don't both have to be hooked up. If you just want to record a mono sound, connect it to the bottom input. This is the left channel. You can also use a cable or switch box to connect the digitizer to your amiga. This is by no means required; however, if you wish to do this keep in mind a few points:

1. The case ground on the digitizer (the edge of the DB25 connector) must remain electrically connected to the amiga connector.
2. If you use a switch box, make sure all lines are passed bi-directionally. It is recommended you try the switch box with your *Perfect Sound* before you buy it.



Setting your digitizer

The two knobs on the front of your digitizer control the gain of an input amplifier. The output from this amplifier is what is digitized by an analog to digital converter. These knobs will have to be adjusted for different audio sources, or even for different sounds on the same audio source if you desire high quality results. For example, if you are digitizing a whisper you may want to increase the gain of the input amplifier. This will allow you to get a range of values from 0 to 255, thus using the full precision of the A/D converter. This will result in clearer sound. When you play the sound back, it will be louder. However, just lower the volume to get the whisper back.

To adjust these knobs for a source there are two methods that can be used. The easiest is to select the “*Monitor*” item under the “*digitize*” menu. With an audio source connected, adjust the knob at the front of the digitizer to increase the volume. The idea is to get the sound output as loud as possible without introducing distortion. Distortion will occur when the gain of the input amplifier causes the signal levels to pass out of the range of the A/D converter.

The second method is to pick the “*Monitor Signal Levels*” submenu option. This will display actual numbers that the A/D converter is producing. In theory, you would want to get the maximum value so it hits 255 only occasionally, and the minimum value to hit 0 just as rarely. In this way you are using the full range of 0 to 255, but not going out of this range, which will result in distortion. However, in practice, this method does not work too well, and a slightly higher gain is usually better. Try a combination of both methods until you get something that sounds good.

An easy way to check your input gain is to record a sample, and then graph it. If you see a lot of flat peaks, your gain is probably too high. See the “*Graph current range*” option under the “*Special*” section.

If you are recording in stereo, you need to have the input gain of both channels identical. To help you with this, the monitor signal levels submenu displays the difference between the right and left channels. Although it is impossible to get this number to stay at zero, the smaller it is the better.

Audio Sources

Almost any audio source can be used with *Perfect Sound* — stereos and VCRs are perhaps the most common. Connect these two devices to the digitizer with a patch cord (a shielded cable with two male RCA jacks on each end.) *Perfect Sound* can also be connected to speaker jacks on the back of radios or cassette players. These sources, however, might not digitize as well as a line level output, as in a VCR audio out. You may need to purchase a mini—plug to RCA adapter for some audio sources. You can find these adapters at most electronic stores. Note that a microphone can not be plugged directly into the digitizer. If you wish to digitize from a microphone, you need to go through a pre-amplifier. A small mini-amp will work. Other possibilities are small battery equalizers, or even your stereo. Any device that takes a mic input and a line or external speaker output will work.

Basic Sound Editor Functions

Once *Perfect Sound* is loaded, you will notice six gadgets — three sliders, a small speaker, a stop button, and a squiggly line (this is the “graph range” button.) The three sliders are marked *start*, *end*, and *pos’n* (position). These adjust your start of range marker, end of range marker, and insert position. The sample that these markers are altering is highlighted at the top of your screen. While you are changing these markers, clicking the right mouse button will play the current sample up to the marker. By using the start and end markers, you define a range that editing commands work on. For example, you can define a range, then delete that range by selecting the “*Delete Marked Range*” option in the “*Edit*” menu. On the right of each slider are two arrows. These are used to give you precise control over the position of the slider. By clicking these arrows, you can increase or decrease a value by one. These are particularly useful for editing instruments.

The other gadgets are in the lower right corner of your status area. By clicking these gadgets, you can play the current marked range, stop the all sound, or graph the marked range.

When you load or record a sample, it will appear in the upper half of your display. Up to 15 samples can be in memory at the same time. The current sample (the sample that the start and end sliders adjust) will be in inverse video. To choose a new current sample, you simply click its name with the mouse. Double clicking a name will play the sample. To try this out, go to the “*Project*” menu and select “*Open.*” Change the directory to “*sounds*” by clicking “*sounds*” when it appears in the “*Load Sound*” window. After you are in the “*sounds*” subdirectory, click a sound in the window to select it. Then click “*ok*” to load it.

As well as digitized sounds, you can load digitized instruments such as from “*Instant Music.*” When you have loaded an instrument file, you can play the different octaves by using the function keys. Each key, starting with F1, plays a different octave from the instrument.

Project

New

This will delete all your current samples.

Open

Use this to load a sample. The editor will automatically recognize IFF format. If the file format is not recognized, the file will be loaded as raw data. If this is the case, no playback speed will be available and you will have to set it.

Selecting the open menu will bring up the disk I/O requester. The current drawer or directory will be displayed. Use the slider on the right side to move around in the directory. To select a directory or file, just click it. You can also click the file and drawer gadgets below the file names in order to type in a new name directly. To select the internal drive root directory, enter df0: and to select the first external drive, enter df1: in the drawer box. Once you have selected the desired file and directory, click the “ok” box at the bottom of the requester.

Save As . . .

This option saves the current sample to the current directory. Type the file name into the disk I/O requester. Use the submenu to pick which format to use — IFF, DUMP or COMP.

IFF saves the file as a Form 8SVX in an IFF file, the standard format for sound on the Amiga. Most programs, such as *Deluxe Video* and *Instant Music*, expect IFF files.

DUMP will save only digitized sound data with no other information. Stereo sound will save only one channel. If you wish to dump stereo sound, break it up into two samples (see edit menu) and save each sample separately.

COMP saves the file in IFF format using *Fibonacci Delta Compression*, a technique that stores each sample as a 4 bit offset from the previous sample. This will cut your file size in half, but may reduce the sound quality considerably.

Save

This option saves the current sample using that sample name as the file name. Pick which format to use with the submenu.

Quit

This exits Perfect Sound.

The Edit menu

Delete Marked Range

After you have set start and end markers, use this submenu to delete the range. There is no way to recover a deleted area.

Discard this Sample

This option removes the currently highlighted sample from memory. Deleted samples can not be recovered.

Insert marked range

This option will insert part of one sample into another or even the same sample. First select the sample that contains the sound you wish to copy. Use the end and start gadgets to select the desired range. Now select a new sample that you wish to copy the previously marked range into. Move the position gadget to the point where the marked range is to be copied into. Now pick the “*Insert marked range*” menu. You will be asked to click the sample where the marked range is to come from. Point at the sample you had previously marked a range in, and click the left button. The marked range from the sample just clicked will be copied into the currently highlighted sample at the position marked by your pos’n gadget.

Change Playback Period

Use this submenu to change the speed at which the current sample will be played at. The speed you enter is in “*samples per second.*” One use for this is to change the sex of a voice. For example, a male voice will sound female if it is sped up.

Once you select this option, a requester will appear with a proportional gadget and a number. You can adjust the slider gadget to change the number, or click the number and enter a value directly. The slider will have a limited range, but by clicking the number and typing in the value, any number can be entered.

Note that the Amiga hardware is limited to a maximum playback speed of 28,876 samples per second.

Copy Range to New Slot

This option creates a new slot with the currently marked range. An easy way to edit is to break up a sample into a lot of little samples — one word per slot for example. Then use the “*Append Slot to Slot*” option to put them back together in any order.

Append Slot to Slot

This option allows you to append any two slots together. You will be asked to click the slot to append, and then to click the slot to append it to. Just follow the prompts.

Create Stereo

This option allows you to merge two mono sounds into a stereo sample. You will be asked to point to and click the slot for the right and left channels.

Break up Stereo

This option create two mono samples out of the two channels in a stereo sample.

Special

Flip this Sample

This submenu will reverse the current sample. If you now play the sample, it will play backwards. This is useful for listening to your Led Zeppelin records in reverse.

Graph Marked Range

This will draw a graph of the sound between the start and end markers. For large samples, the graph will lose realism since each point plotted represents a large number of actual points. By moving the markers, however, you can zoom in on a point to get a truly accurate graph. Two horizontal lines above and below the graph show the minimum and maximum values possible. If your waveform hits these lines often, it may indicate distortion due to the input signals going outside the range of your digitizer. Lowering the input gain with knobs on the front of the digitizer will help. See the section “*Setting your digitizer*”.

When you graph a range, the first and last samples will be printed at the bottom of your screen. These numbers are the digitized value of the first and last points of the displayed waveform. This information is useful for connecting samples “smoothly.” Also, these numbers are helpful in creating instruments. The repeating waveform should start and end with values that are close to each other. This is to avoid “clicks” that occur with a sudden jump from one value to a different value.

Create instrument

This option is used to create an instrument in IFF format for use in music editors such as “*Instant Music.*”

When you pick this option, your octaves (which should have already been created in separate slots) will be merged into a new instrument. The octaves will be sorted in order of decreasing frequency, and you will be warned if any octaves have a length that is not twice the previous length.

Note: to play the notes of an instrument, use the function keys at the top of your keyboard. You can also load an instrument from *Deluxe Video* or *Instant Music* and play it with the function keys. You should make it a point to play any instrument you have just created with the function keys. This is because you will be warned if the instrument you just created does not conform to the IFF specs when you press the function key.

An instrument file consists of one or more octaves. Each octave has two sections, a repeating part and a non-repeating part. The non-repeating part of a sound is the first part of the sound, called the attack — for example, pressing the key on a piano causes an initial sound. This is the non-repeating part.

Following the non-repeating part of an octave is an optional repeating part. This is sampled data that can be played over and over to create a continuous tone. For example, when you blow into a flute, there is an initial sound (the non repeating part. Then, as long as you blow at the same rate there is a continuous tone...this is the “repeat part.”) Some instruments, such as a drum, have no repeating part. Most instruments will have both parts.

The IFF file format requires that the octaves of an instrument be in order of decreasing frequency. Each successive octave needs to be twice as long as the previous octave. The repeating portion of each octave within an instrument should have the same number of cycles.

For a technical discussion on the IFF instrument format, see the file on this disk called “*FORMAT_IFF.*” This is information on IFF instrument files as put out by Electronic Arts and Commodore-Amiga.

To create an instrument with *Perfect Sound*, do the following:

1. Clear all slots of all samples.
2. Digitize (or load) a sound that you wish to use as your instrument.
3. You will put each octave in a separate slot. The easiest way to create another octave is to use “*Freq = Freq*2*” to create a higher octave or “*Freq = Freq/2*” to create a lower octave.

Thus:

- a. After step 2 above, select “*Freq = Freq*2*” from the “*Special*” menu.
- b. Click on your original sample to select it.
- c. Select “*Freq = Freq/2*” to create another octave.

You now have three octaves. Play each octave to hear them.

4. After creating each octave in a separate slot, click the octave with the highest frequency to select it. Now move the start of range marker to indicate where the repeat portion of the octave starts. If there is to be no repeat part, move the start marker to the far right.
5. Save your octaves separately — an instrument can not easily be broken up into its octaves.
6. Pick the menu option “*Create Instrument.*” Use default values for any questions you are asked. The result is your IFF instrument. You can play it with your function keys, then save it to disk. You should be able to load it with *Instant Music* or any program that uses IFF instruments.

*Freq = Freq*2*

This command creates a new sample with twice the frequency and half the length of the current sample. This is done by removing every other sample from the current sample.

Freq = Freq/2

This function takes a sample and divides its frequency by two without changing the playback period. This is done by doubling the sample size and creating new samples by interpolating between existing samples.

This command is useful for creating instruments. You can digitize one octave, then create other octaves using the commands “*Freq = Freq/2*” and “*Freq = Freq*2.*”

Auto Graph

Use the submenu to turn “*auto graphing*” on or off. When “*auto graph*” is on, the graph will be updated every time you change samples or adjust the start or end markers.

Zero Range

This will cause all samples between the markers to change to zero.

Digitize

Alter Record Speed

This option allows you to enter a new recording speed. Enter the value for the desired playback speed. A smaller speed results in poorer sound quality but uses less memory.

The software can only record at discrete intervals, so the speed you get will not be exactly what you request, but will be as close as possible rounded down. The values you enter are in “*samples/second*”

Note: A good default value is 10,000 samples per second. This results in fair sound, and uses 10K of memory per second for mono sound. Also, the Amiga has a low pass filter on its output with a -3db point at 4 KH and completely eliminates frequencies above 7KH. Standard sampling theorems state that you need to sample a sound at twice the desired response frequency. Thus a sampling rate of 14,000 samples/second is the theoretical maximum you would want to use.

Monitor digitizer

This option will cause *Perfect Sound* to play the current input sound. Nothing is being recorded, the sound is just being passed through from the digitizer to the Amiga's sound channel. Use the submenu to pick which channel should be used (Right, Left, or Stereo).

Record a Sample

This option will put you in monitor mode. However, as soon as you click the left mouse button, you will start recording. Click the left mouse button again to halt recording. Use the submenu to select which channels to use.

View Signal Levels

This option will display the actual numbers being received by the analog to digital converter. You will see the minimum received so far, the maximum received so far, and the actual value. These numbers are displayed for both the left and right channels. Also, the difference between the two channels is displayed. You can use the numbers for adjusting the knobs on the front of your digitizer. See the section *“Setting your digitizer”* for a complete discussion on how to adjust your digitizer. Note that silence is not received as a zero. A zero input voltage (silence) will digitize to 128 (ideally). A number smaller than this represents a negative voltage, and a larger number represents positive voltage. This is different than the values used by the Amiga’s digital to analog converters. These converters use zero for silence, negative numbers for negative voltage, and positive numbers for positive voltage. The software makes the conversion when a sound is recorded.

Notes
