Digimaster

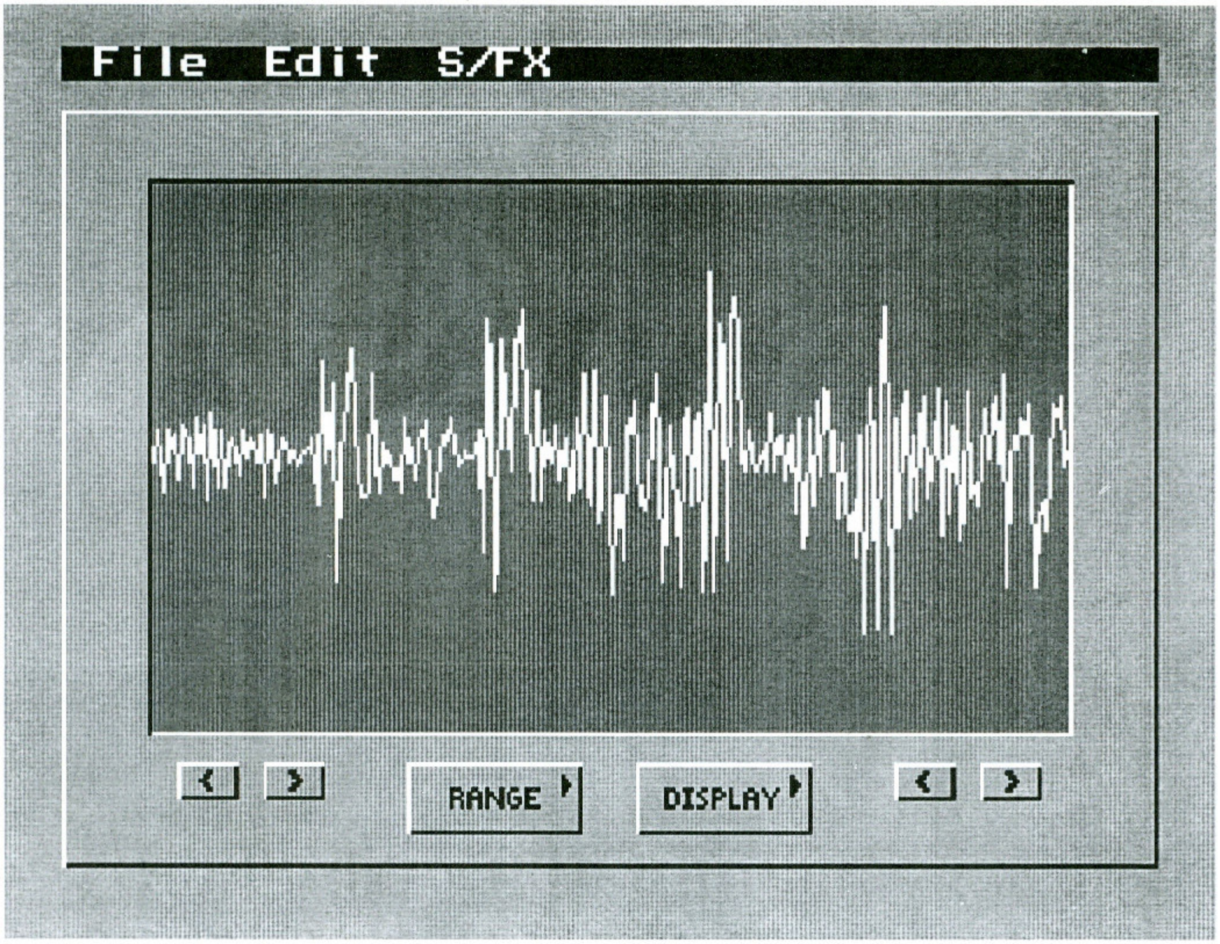
User Manual

Autumn Technologies

Digimaster

Digital audio sampling and editing software for your Commodore 64

Autumn Technologies



Digimaster is a unique software product which allows you to process digital audio on your Commodore 64.

This advanced software utilizes a fully graphical interface with pull down menus. Incorporated into it's operation are many powerful features such as Cut, Copy, and Paste, which makes editing sound as easy as editing text in a word processor. Using an optional audio digitizer, live sound can be grabbed into the computer, edited, and then saved to disk. There is even a utility included which will convert Amiga sound samples, giving you access to a vast library of sounds.

Probably the most impressive feature of this software is it's ability to replay sounds in true eight bit digital audio on your Commodore 64 without the need for extra hardware. This is made possible by a revolutionary method of controlling the sound chip inside the Commodore 64. The result is crystal clear audio reproduction.

Digimaster Features:

- Fully graphical interface
- Many powerful editing functions
- Ability to replay sounds in true eight bit digital audio without the need for extra hardware
- Included software which allows replaying sounds from your own programs

Requirements:

Commodore 64 computer
Commodore 1541 or compatible disk drive
Joystick or mouse (mouse highly recommended)

Acknowledgements

Many thanks go to all the people who have been instrumental in the development of this product. I would especially like to thank:

Jim Sutherland and Brian Richter, owners of Applied Computer Systems, for supplying technical manuals I didn't have, and letting me work part time at the store so I could pay my bills.

Brian Smither, president of the Sacramento Commodore Computer Club, for all his good advice, and the use of his computer equipment.

Jack Vander White, owner of CEE-64 Alive disk magazine, for the use of his European C-64 which aided in the development of the PAL version of Digimaster.

Doug Cotton of Creative Micro Designs for all his input.

Audio Sampler for Digimaster

This sampler is warranted for use with Digimaster only for 30 days from purchase AND is not guaranteed to work with the "low sampling rate" of Digimaster (it may work but only with some C64s). This digitizer will perform normally with the High and Medium sample rates. With the low sample rate your computer may lock up. In testing this product it was discovered that the low sample feature only worked on some computers and no pattern was noted to predict which would and would not work. I recommend that you not use the low sample rate with this product.

Instructions:

TURN OFF your computer and all disk drives and monitors BEFORE you plug the AUDIO Sampler into the User I/O port of your C64 computer. FIRST after plugging in the Audio Sampler turn on your monitor then the computer. IF YOUR COMPUTER DOES NOT START UP NORMALLY TURN IT OFF! Your computer should start normally and the familiar blue screen appear in the usual amount of time (15 to 20 seconds) if this does not occur turn off your computer and check the connections.

AFTER the above test is done and the computer is working normally boot Digimaster.

It is a good idea to have a formatted blank disk ready to save your samples. 1541 drives seem to work the best, I have had problems with Digimaster and a 158l.

The female RCA plug in the Audio sampler is for the audio input, use only battery powered devices for input. Any normal audio out from your tape player or low power amp should work. The warranty applies ONLY for use with battery powered devices. MAKE SURE that the volume on your INPUT is turned very low! Set your input volume to the first 10% of the volume range or less. (You will adjust the volume when sampling.) High volumes will produce very poor quality samples and can damage the sampler.

Follow the directions that came with Digimaster to sample your audio inputs. After you sample a sound the screen will display a wave analog of that sound. If the "wave" is clipped at the top and/or bottom of the display, you had the volume too loud and the playback will be of poor quality. Your goal is to have only a few peaks hitting the top or bottom of the display; that will give you the cleanest sample at the highest volume on playback.

On the TOP of the AUDIO sampler there is a hole. Inside the hole is an adjustable resistor for trimming the AUDIO sampler to your input device. Adjust this if necessary while you are listening to the sound coming out of your computer during the sample process (when you choose Sample Audio from the S/FX menu you will here the sound coming from you computer speaker, you click the mouse or joystick to sample after you adjust the volume of your audio input and the Audio sampler trim if necessary.) The trim was adjusted before the audio sampler was sent out but may require adjusting to your system but do this only after you are unable to get a clean sound by adjusting the volume of your audio input device.

To adjust the “trim” on the Audio sampler use a small screw driver and gently turn the trim as you listen till you get the best possible sound. BE CAREFUL not to touch other things inside the Audio sampler. The schematic for this device was published in Commodore World magazine Issue number 3 Volume 1, Number 3. The only change is the R9 is too small as published so the unit you have has more than 910 ohms at this location. This is included in case you may some day need a repair.

If you enjoy your Audio sampler PLEASE, recommend it to your friends. Digimaster is a great program and a real joy to sample with.

Welcome to Digimaster

Digimaster is a sampling and editing software package which will let you process eight bit digital audio on your Commodore 64. The Digimaster package includes an editor, which when used with an audio digitizer, allows you to capture live audio into your computer. This audio sample can then be edited and saved to disk. The package also includes software which will let you use these sound files in your own programs.

Getting Started

The first thing you should do is make a copy of the program disk. NEVER USE YOUR ORIGINAL. Once you have a working copy, you must configure Digimaster. Load and run the program named SETUP. You will be asked whether you want to use a joystick or a mouse. Type in J if you will be using a joystick, or M if you will be using a mouse. You may also want to format a disk at this time. This disk can then be used to hold the temporary files used by Digimaster.

The Editor

The Digimaster editor is a tool which allows you to bring sounds into your computer, and tailor certain attributes of those sounds to meet your specific needs.

One of the features of the editor is to capture live sound. To do this you will need an audio digitizer. There are a number of good audio digitizers available for the Amiga. An interface cable is required to use one of these Amiga digitizers with Digimaster. Refer to the section on the interface cable for more information.

Another feature of the Digimaster editor is the ability to edit sounds that are already stored as disk files. This package includes a conversion utility that will convert Amiga sound files into Digimaster sound files. For more information on this subject, refer to the section on the ConvertIFF utility.

The editor gives you a visual representation of the audio data stored in memory by displaying a waveform in the scope window. This waveform represents the entire audio sample in memory, and since the amount of data in memory will vary between audio samples, this waveform has no fixed scale. It's simply a graphic representation of the entire audio sample.

Loading the Editor

To load the editor from drive 8, put the disk containing the Digimaster files into drive 8 and type LOAD "DIGIMASTER",8,1. This will load and run the editor. If you have the files on a disk in a

different drive, use that drive number for the LOAD command. Note also that if you are using a Commodore 128, you will have to be running in 64 mode.

Using the Mouse

The Digimaster editor works best with the 1351 mouse. If you don't have one, it is highly recommended that you purchase one. Most operations involve using the left mouse button. The right mouse button is used primarily for accessing the menus. The mouse should be plugged into port 1.

Using the Joystick

When using the joystick, the fire button is used to access the menus as well as most other editor operations. The joystick should be plugged into port 2.

The Cursor

The cursor appears as a thin vertical line in the scope window. Move the pointer anywhere inside of the scope window and then press and hold down the left mouse button or joystick button. You will see the cursor appear. This sets the cursor position. Now let go of the button. The cursor will disappear, but the cursor position will still be set at that point in the scope window. Some of the editor functions require you to first set the cursor position.

Marking a Range

Many operations in the editor involve marking a range in the scope window. To mark a range, move the pointer to the desired position in the scope window. Press and hold down the left mouse button, or fire button on the joystick. You will see the cursor appear. While holding down the button, move the pointer to the right, and continue moving the pointer until you have marked as much of the data as you wish. Now you can let go of the button.

If you need to make minute adjustments to the size of the marked range, you can use the small buttons on the scope panel. The small buttons on the left side of the scope panel are used to adjust the left side of the marked range in one pixel increments. It should be noted that when you change the left side of the marked range you also change the cursor position. The small buttons on the right side of the scope panel are used to adjust the right side of the marked range in one pixel increments. You can clear the marked range by clicking once anywhere in the scope window. Of course, this also sets the cursor position to the position where the pointer was when you cleared the range.

Requestors

Requesters are boxes which appear on the screen and are related to certain editor functions. There are three types of requesters used in the Digimaster editor. Message Requesters, Selection Requesters, and the File Requester.

Message Requesters simply inform you of certain events. When you are finished reading the message, press the left mouse button or fire button on the joystick. Selection Requesters allow you to make a selection for a certain editor function. These requesters will always allow you to cancel if you didn't want to perform that particular operation. Click on the checkmark to proceed with the selected operation, or click on the X to cancel.

The File Requester appears for operations involving loading and saving files. When the File Requester comes up, the first ten files in the directory of the active disk will be shown in the file requester's directory window. If there are more files in the directory, you can click on the MORE button to see them. To see the directory again, click on the REREAD button. If you want to access another disk, click on the DISK button. A message requester will appear informing you to insert a disk. Press the left mouse button after inserting the new disk. If you have more than one disk drive, you can click on the DEVICE button to go to the next disk drive. To select a file, click on that file's name in the directory window. The filename will appear in the box below the directory window. If you wish to type in a name, as you would if you were saving a new file, click in the box under the directory window. End your typing by hitting RETURN.

Disk Drives

When Digimaster first starts up, it remembers which disk drive it was started from and uses that drive for all disk files. Digimaster allocates all available memory in the computer for the digital audio data, and since there is no memory left over, the disk drive is used for storing temporary data. If you wish to use a different disk drive for these temporary data files, choose CONFIGURATION from the EDIT menu, and change the drive number for the CLIP DRIVE.

Playback

The two large buttons in the middle of the scope panel allow you to listen to the sound sample displayed in the scope window. The button marked DISPLAY will play the entire sample. The button marked RANGE will play the portion of the sample that is marked in the scope window.

Menus

To select a menu item with the mouse, move the pointer to the menu heading of your choice, then press and hold down the right mouse button or fire button on the joystick. You should see the menu appear. Now move the pointer to the menu item you wish to select and let go of the button. Below is a description of the menu functions.

File

Load: Use this to load sound files into the editor. You can only load DFF files. DFF is an acronym for Digimaster File Format. Provided is a utility to convert Amiga IFF files to DFF files.

Save: Use this to save all the data in the scope window as a OFF file. This operation will put up a requester asking if you wish to overwrite the current file. Click on the checkmark to overwrite the file, or click on the X to cancel the save. If you haven't yet loaded a file during this session, the File Requester will appear allowing you to enter a name for the sample you are saving.

Save As: Use this to save all the data in the scope window as a OFF file under a new name. When this function is selected, the File Requester will appear. Here you can type in the name you wish to use for this sample. If you try to use the name of an existing file, you will be asked if you wish to overwrite that file. Click on the checkmark to overwrite the file, or click on the X to cancel the save.

Save Range: Use this to save the marked range in the scope window as a OFF file. If you try to use the name of an existing file, you will be asked if you wish to overwrite that file.

About: This will put up a requester which displays a copyright message and a version number.

Quit: This will return you to BASIC. Always make sure you have everything saved before quitting.

Edit

Copy: This will copy the marked range into the clip file. This data will be used for the paste functions.

Cut: Same as Copy except the marked range is deleted from the scope window after being copied to the clip file.

Erase: This will delete the marked range without affecting the data in the clip file. Careful, once you use this, that data is gone.

Paste Over: This will try to mix the data being held in the clip file with the data in the scope window. The paste will begin at the cursor position. If there is not enough memory to paste the clip data, the operation will be aborted. This function can be used to create echo effects, or to mix a number of different sounds.

Paste Insert: This will try to insert the data being held in the clip file into the data shown in the scope window. The point of insertion is at the cursor position. If there is not enough memory to add the clip data, the operation will be aborted.

Configuration: This will allow you to select the disk drive used for loading and saving DFF files as well as the disk drive used for temporary files such as the clip file. Here, you can also adjust the

point at which the filter cuts off the high frequencies during playback. If you have an older computer, the default setting may cause playback to sound muffled. If this happens, try raising the cutoff point until the sound is clear. If the sound is harsh or distorted, try lowering the cutoff point. For more information on this, refer to the section on sampling tips.

S/FX Volume: This allows you to change the volume for a marked range. The START% value determines the percentage at which the volume operation will begin. The END% value determines the percentage at which the volume operation will end. The volume operation always occurs from left to right in the scope window. One use for this is to create a fade-in or a fade-out.

Resample: This will alter the sample rate for the entire sample in the scope window. If raising the sampling rate, this function will use a temporary file on the clip drive. Make sure there is plenty of room on the disk. This function can take quite

Backwards: This will flip the marked range of data around so that it is backwards. I don't know what practical use this would have, but it's there for anyone who wants to use it.

Loop Range: This will play the marked range in a continuous loop. Hold down the right mouse button, or hold the joystick forward to stop playback.

Sample Audio: If you have a digitizer hooked up to the user port, this will allow you to bring live sound into the computer. When you select .this option, you will be prompted to select the sample rate you wish to use. Refer to the section on sampling tips for information on which sample rate to use. Click on the checkmark to monitor the incoming sound. The screen will turn dark grey and you should be able to hear the sound that is being fed into the digitizer. This will allow you to adjust the attributes of the incoming audio before actually sampling into memory. Press the left mouse button or fire button on the joystick to begin sampling the sound into memory. The screen will turn medium grey to indicate that the sampling process is in effect. When you wish to stop sampling, press the left mouse button or fire button on the joystick. Also, the sampling process will stop automatically when all the memory in the computer has been used. You will now see a waveform in the scope window that represents the sound that was just sampled into the computer.

Error Messages

From time to time you may encounter error messages. What follows is a description of some of the possible causes for the various error messages you may see.

Out of Memory

The sample you are trying to load is too large to fit into the block of memory allocated by Digimaster.

The operation you are trying to perform requires more memory than is available.

File Error

The sample you are trying to load is not in a format that is recognized by Digimaster.

Disk Error
A read or write error has occurred during disk access.

An invalid file name has been used for loading or saving.

The disk is full.

ConvertIFF

Amiga sound samples are stored as IFF files. IFF is an acronym for Interchange File Format. This format is not directly usable by Digimaster. ConvertIFF will create a Digimaster DFF file from an Amiga IFF sound file. Upon running ConvertIFF, you will be prompted to enter the drive number for the disk drive that contains the disk with the Amiga IFF sound file. Press RETURN to use the same drive that ConvertIFF was loaded from. You will then be asked to enter the name of the Amiga IFF file you wish to convert. Press RETURN to see a directory of the disk. After entering the file name, you will be asked to enter the sample rate that the Amiga IFF file will be converted to. Enter L for Low, M for Medium, or H for High, or simply press RETURN to use the Low sample rate. After reading the Amiga file into memory, you will be asked which drive to save the sample to. Press RETURN to use the same drive. Here, you can also insert another disk to save the sample to. Next, you will be asked to enter a file name for the new sample. Press RETURN if you wish to see a directory of the disk.

PlayFile

With this utility, you can listen to Digimaster files without having to load up the editor. When this program is run, it will try to load the player program. If this file cannot be found, PlayFile will exit with a FILE NOT FOUND error. Once this program has been loaded you can put in the disk that contains the sound file you wish to play. You will then be prompted to enter the Filter Cutoff. Enter a number between 0 and 255. You may have already determined which value produces the best sound on your system. If you previously entered a value here, you can simply hit RETURN to keep that value. Next, you will be asked which drive to load the sound file from. Press RETURN to use the same drive. When prompted for the file name, enter the name of the sound file you wish to hear and press RETURN. After the sound has played, you can press RETURN to hear it again, or any other key to continue. You will be asked if you wish to play another sound. If so, enter yes, or simply y. If not, enter no, or n.

Tips on Sampling

Digimaster uses three different sampling rates. Low, Medium, and High.

The low sampling rate will provide the most time, but will deliver the poorest sound quality. One benefit of using the low sample rate is that the screen is not blanked during playback. However,

some things will just not sound good when using the low sample rate. Generally, this rate is only good for human voice.

The medium sample rate yields better sound quality, but allows for less time. Using this rate causes the screen to be blanked.

The high sample rate provides the best sound quality, but allows for the least amount of time. The screen is blanked during playback of samples using the high sample rate.

One caveat of digital audio is a phenomenon known as aliasing distortion. This distortion can be heard as a high pitched whistling or screech, and occurs when the original sound contains frequencies that are higher than the sample rate. The best way to avoid this is to filter out these higher frequencies before the sound gets to the digitizer. Some of the Amiga digitizers have a filter built in to solve this problem. Another way to avoid aliasing distortion is to filter out the higher frequencies during playback. This is what the filter cutoff is used for in Digimaster. This doesn't really get rid of the aliasing, it just makes the distortion caused by the aliasing less audible. One point to -remember is that aliasing will decrease as the sample rate increases.

A good method for sampling is to first record the sounds onto cassette tape. When recording to tape, use a Normal Bias tape (also known as Type I). On the tape deck, set the Bias to Chrome (also known as Type II) or to Metal (also known as Type IV), depending on how much of the higher frequencies you want filtered out. Setting the Bias to Metal will have the most noticeable effect. When playing the tape back into the digitizer, you can set the playback equalization on the tape deck to 70us (Chrome; Type II, Metal; Type IV) to filter out even more of the high frequencies. Filtering the higher frequencies will effectively reduce aliasing. It should be noted that some of the newer cassette decks will automatically adjust to the type of tape you put in. Therefore, this method will not work with such tape decks.

Samples will always come out better if the volume of the sound going into the digitizer is as high as possible without causing distortion. If you have a way of adjusting the volume of your audio source, such as an output level control on your cassette deck, try bringing the level up until it starts to sound distorted. Then back it off slowly until the sound is clear. Of course, you should do this when Digimaster is in monitor mode, not when it is actually sampling the data into memory. Some of the Amiga digitizers have an input level control that can be used to adjust the volume of the incoming sound.

Audio Digitizers

There are a number of audio digitizers available for the Amiga. Most of them are designed to plug into the Amiga's parallel port. These digitizers can be interfaced to the Commodore user port through a simple adapter cable. There are a couple of points to consider when choosing an Amiga digitizer though. A lot of the Amiga digitizers on the market are stereo, and some of them require special software control in order to function. These cannot be used with the Digimaster editor. Only monophonic generic digitizers can be used with the Digimaster editor. The term generic refers to a digitizer which does not require special codes sent to it in order for it to function.

The digitizer used throughout development of Digimaster was the Master Sound digitizer manufactured by MicroDeal. If there is enough of a demand, Autumn Technologies will make available a digitizer made specifically for Digimaster. Until then you can contact your local Amiga dealer for information on digitizers available for the Amiga. You may also wish to contact the manufacturers directly.

Sound Sampler
Phoenix Microtechnologies Pty., Ltd.
18 Hampton Road
Keswick, South Australia 5035
(011) 618-293-8752

Master Sound
MicroDeal
P.O. Box 68
St. Austell, Cornwall PL254YB, England
011-447-266-8020

Interface Cable

The Amiga parallel port digitizer connects to a Male DB-25 connector. The Commodore ~ser port requires a 24 pin edge card connector. Below is a diagram detailing the construction of an interface cable for connecting an Amiga digitizer to the Commodore user port. Refer to Commodore's documentation on the user port for information on pin assignments. If there is enough of a demand, a pre-made cable will be made available for sale. You must read the disclaimer in the back of this manual before attempting to construct this interface cable.

Amiga DB-25	Commodore 24 pin edge	Description
1--------------------8		PC
2--------------------C		Data0
3--------------------D		Data1
4--------------------E		Data2
5--------------------F		Data3
6--------------------H		Data4
7--------------------J		Data5
8--------------------K		Data6
9--------------------L		Data7
14--------------------2		5V
17 through 25------A, 1, N, 12		Ground

Technical Information

Sample Rates

As mentioned earlier, Digimaster uses three sample rates: Low, Medium, and High. The sample rates are based on the processor clock of an NTSC Commodore 64. The Low sample rate has a period of 128 clock cycles, which works out to 7969 samples per second. The Medium sample rate has a period of 100 clock cycles, which works out to 10200 samples per second. The High sample rate has a period of 82 clock cycles, which works out to 12439 samples per second.

Digimaster File Format

The file format used by Digimaster provides information about the audio sample in a header, which comprises the first eight bytes of the file. Actually, since DFF files are stored as PRG files, the first two bytes of the file determine the load address, and the next eight bytes would be the header information. since the two byte load address is actually a mechanism used by the Commodore DOS, this discussion will treat the file as if it started with the header information. The header description is provided in standard 6502 assembler notation.

```
Header    .byte  $04,$00,$00,$00    ;Data Length
          .byte  "DFF"              ;Format ID
          .byte  $80                ; Sample period
Data      .byte  $01,$02,$03,$04    ;Sample data
```

In this example, the header describes a sample that is four bytes long and uses the Low sample rate.

The Length field is a 32 bit number in low byte through high byte format and indicates the number of bytes of eight bit audio data in this sample.

The format identifier is always the three ASCII characters DFF.

The sample period byte contains one of three unique values: $80 for the Low sample rate, $64 for the Medium sample rate, or $52 for the High sample rate. These values indicate the sample period in clock cycles.

The audio data starts with the first byte immediately following the sample period byte. This data must not contain any bytes that have a value of zero. A zero byte always indicates the end of data. When converting audio samples from other formats, any bytes encountered that have a value of zero should be converted to a value of one.

Programming Information

The information contained in this section is provided for personal use only. Any software you write which plays Digimaster sound files may not be distributed in any way without written permission from Chris Brenner and Autumn Technologies. See the section on Developer Information if you are interested in releasing a program which incorporates the Digimaster playback technology.

64Player and 128Player

The 64Player and 128Player are machine language routines which you can incorporate into your own BASIC or machine language programs. These routines are fully re-locatable, which means they can be loaded anywhere in memory and still work correctly. Neither of these routines check to make sure they are trying to play Digimaster files. That responsibility lies with the programmer. The player routines interpret a zero byte as the end of data. It is the programmer's responsibility to properly terminate the DFF file after it has been loaded into memory.

Both the 64 player and the 128 player require that the SID chip be properly initialized first. See the section on the 64 and 128 BASIC routines for information on setting up the SID chip.

The 64 player requires a pointer to first byte of the header for the Digimaster file being played. This pointer must be stored in locations 251 ($FB), and 252 ($FC) in low byte, high byte format prior to calling the player routine.

The 128 player must be loaded into BANK 1, and the Digimaster file being played must also be in BANK 1. As with the 64 player, the 128 player requires a pointer to the first byte of the header for the Digimaster file being played. This pointer must be stored in locations 250 ($FA), and 251 ($FB) in low byte, high byte format prior to calling the player routine.

64Routines and 128Routines

The two files, 64Routines and 128Routines, contain a collection of BASIC routines for playing Digimaster files back in your own BASIC programs. These include BASIC routines for setting up the SID chip, allocating memory for and loading the machine language player, allocating memory for and loading Digimaster files, and calling the machine language player routine to actually play the sample. The routines which load Digimaster files do no checking for the file type. It is the programmer's responsibility to make sure the file being loaded is actually a Digimaster file. These routines are fairly simple and can either be incorporated into your own BASIC programs, or can serve as examples for writing your own BASIC routines.

The routine which sets up the SID chip is very important. This must be called once at the beginning of your program. If the SID chip is not properly initialized, the machine language player routine may not deliver the expected results.

Developer Information

Software authors are encouraged to use the Digimaster playback technology in their programs. If you are planning on releasing such a product, contact Autumn Technologies for detailed programming information and technical information describing the theory of operation of the Digimaster playback routines.

Autumn Technologies, P.O. Box 161805, Sacramento, CA 95816

Digimaster
Version 1.1 Addendum

Getting started with version 1.1

As mentioned in the manual, you should make a backup copy of the Digimaster disk, and then put the original disk in a safe place. Before you can use Digimaster, you must first run Setup. Note that this is a one time operation, and that there is no need to re-configure Digimaster every time you want to use it. See the manual for instructions on how to use Setup.

New features

When sampling, the screen now turns blue to indicate monitor mode. This should make it a little easier to determine which sampling mode you are currently in.

The low pass filter is now turned completely off in both monitor mode and sample mode. This not only makes the sound a little more clear in these modes, but also allows you to hear how much aliasing distortion is present during the sampling process.

The cursor position and range are now indicated in the menu bar. The number following the C= depicts the cursor position within the sample, and the number following the R= indicates the size of the currently marked range. Both of these values represent a number of bytes.

Digimaster File Players

The programs 64PlayFile and 128PlayFile are gone. They have been replaced by two new programs named 64Playlt! and 128PlayIt!. These programs require the new player modules which are included on the disk.

The operation of the 64 version is identical to the 128 version. Simply load and run the appropriate program. Once the player module has been found and loaded, a directory of the current disk will be displayed, and a flashing cursor will appear. If there are more files on the disk than what can be displayed on the screen, the cursor will appear directly over the command Next Page. Hit return if you wish to see the rest of the disk directory, or move the cursor down to the name of the file you wish to load and then hit return. If you have more than one disk drive, you can use the command Next Drive to go to the next disk drive.

Once the sound file has been loaded, the word 'Ready' will appear in the status bar at the top of the screen. You can now play the sound by hitting the F1 key. If you wish to load another sound, hit F3. To change the filter cutoff, hit F5 and then enter the new cutoff value. Hit F7 to exit the program.

Typewriter Keys

This is a new program which demonstrates a practical (well, maybe not really practical) use for the Digimaster player module. This program makes your C-64 sound like an old-fashioned typewriter. It requires the 64player module, and the two sound files named Return and Type, so make sure you have a disk containing these files in the drive before running the program.

If all goes well, you will see a message stating that the key sounds have been enabled, and you will of course hear the keys now when you type on the keyboard. Hitting the key combination RUN/STOP-RESTORE will disable the sound. If this happens, just enter SYS679 to enable the sound again.

Player Modules

The ML player modules have been rewritten to be more efficient, and allow for greater flexibility for the programmer. The following pages document the 64 and 128 player modules. The appropriate player modules (NTSC or PAL) are installed on the disk by the Setup program, and are named 64Player, and 128Player. For examples on calling the player modules in your own program, refer to the files 64Rountines and 128Routines.

Customer Support

If you have questions or comments about Digimaster, feel free to contact us by mail or phone.

Autumn Technologies
P.O. Box 161805
Sacramento, CA 95816
(916) 456-2940

64Player(pointer,option)
$fb A

Description

Plays eight bit audio through channel one of the SID chip. The SID chip must be properly initialized before calling this function. Note that the player code is completely position independent, and may be loaded anywhere in memory.

The pointer to the audio data is taken from zero page at $fb and $fc. The first byte of the audio data must contain the period of the playback rate for the audio sample. This period value is obtained by dividing the 6502 clock rate, which is 1.02 MHz, by the desired playback rate. For instance, if your sample rate is 10.2 KHz, the period value can be obtained with 1020000/10200. The period for this sample would then be 100. Note that if the period is 128 then the screen will not be blanked during playback. Valid period values range from 82 to 164.

The audio data must be terminated with a 0 byte. Any bytes in the audio data that have a value of 0 must be converted to a value of 1. This is because the player interprets a 0 as end of data.

The value passed in the A register determines how the player will act on entry and exit:

These special options can be used when you want to playa number of samples one after the other without any breaks in between. When chaining samples, you must use option 1 for the first sample, and option 3 for the last sample. You must also remember to set the address of your sample into $fb and $fc before you call the player each time. Note also that when using any of the special options the screen will be blanked during playback regardless of the sample rate.

During initialization, the player stores the processor status and memory configuration. These attributes are then restored when the player performs cleanup.

Resource Usage

A,X,Y
$02,$9e,$9f,$fb-$ff
CIA #2 Timer B
VIC Raster Register

PAL64Player(pointer,option)
$fb A

Description

Plays eight bit audio through channel one of the SID chip. The SID chip must be properly initialized before calling this function. Note that the player code is completely position independent, and may be loaded anywhere in memory.

The pointer to the audio data is taken from zero page at $fb and $fc. The first byte of the audio data must contain the period of the playback rate for the audio sample. This period value is obtained by dividing the 6502 clock rate on an NTSC C-64, which is 1.02 MHz, by the desired playback rate. For instance, if your sample rate is 10.2 KHz, the period value can be obtained with 1020000/10200. The period for this sample would then be 100. Note that if the period is 128 then the screen will not be blanked during playback. Valid period values range from 82 to 164, and these values are adjusted through a lookup table for proper playback speed on a PAL C-64.

The audio data must be terminated with a 0 byte. Any bytes in the audio data that have a value of 0 must be converted to a value of 1. This is because the player interprets a 0 as end of data.

The value passed in the A register determines how the player will act on entry and exit:

Option 0 – The player will initialize itself on entry and clean-up on exit
Option 1 – The player will initialize itself, but will skip the clean-up process
Option 2 – The player will skip both initialization and clean-up
Option 3 – The player will skip initialization, but will clean-up when finished

These special options can be used when you want to playa number of samples one after the other without any breaks in between. When chaining samples, you must use option 1 for the first sample, and option 3 for the last sample. You must also remember to set the address of your sample into $fb and $fc before you call the player each time. Note that when using any of the special options the screen will be blanked during playback regardless of the sample rate.

During initialization, the player stores the processor status and memory configuration. These attributes are then restored when the player performs cleanup.

Resource usage

A,X,Y
$02,$9e,$9f,$c3,$c4,$fb-$ff
CIA #2 Timer B
VIC Raster Register

128Player(pointer,option)
$fa A

Description

Plays eight bit audio through channel one of the SID chip. The SID chip must be properly initialized before calling this function. Note that the player code is completely position independent, and may be loaded anywhere in bank 1. The sample data must be in bank 1.

The pointer to the audio data is taken from zero page at $fa and $fb. The first byte of the audio data must contain the period of the playback rate for the audio sample. This period value is obtained by dividing the 6502 clock rate, which is 1.02 MHz, by the desired playback rate. For instance, if your sample rate is 10.2 KHz, the period value can be obtained with 1020000/10200. The period for this sample would then be 100. Note that if the period is 128 then the screen will not be blanked during playback. Valid period values range from 82 to 164.

The audio data must be terminated with a 0 byte. Any bytes in the audio data that have a value of 0 must be converted to a value of 1. This is because the player interprets a 0 as end of data.

The value passed in the A register determines how the player will act on entry and exit:

Option 0 – The player will initialize itself on entry and clean-up on exit

Option 1 – The player will initialize itself, but will skip the clean-up process

Option 2 – The player will skip both initialization and clean-up

Option 3 – The player will skip initialization, but will clean-up when finished

These special options can be used when you want to playa number of samples one after the other without any breaks in between. When chaining samples, you must use option 1 for the first sample, and option 3 for the last sample. You must also remember to set the address of your sample into $fa and $fb before you call the player each time. Note that when using any of the special options the screen will be blanked during playback regardless of the sample rate.

During initialization, the player stores the processor status and memory configuration. These attributes are then restored when the player performs cleanup.

Resource Usage

A,X,Y
$92,$96,$9b,$c3,$c4,$fa-$fe
CIA #2 Timer B
VIC Raster Register

PAL128Player(pointer,option)
$fa A

Description

Plays eight bit audio through channel one of the SID chip. The SID chip must be properly initialized before calling this function. Note that the player code is completely position independent, and may be loaded anywhere in bank 1. The sample data must be in bank 1.

The pointer to the audio data is taken from zero page at $fa and $fb. The first byte of the audio data must contain the period of the playback rate for the audio sample. This period value is obtained by dividing the 6502 clock rate on an NTSC C-64, which is 1.02 MHz, by the desired playback rate. For instance, if your sample rate is 10.2 KHz, the period value can be obtained with 1020000/10200. The period for this sample would then be 100. Note that if the period is 128 then the screen will not be blanked during playback. Valid period values range from 82 to 164, and these values are adjusted through a lookup table for proper playback speed on a PAL C-128.

The audio data must be terminated with a 0 byte. Any bytes in the audio data that have a value of 0 must be converted to a value of 1.

The value passed in the A register determines how the player will act on entry and exit:

Option 0 – The player will initialize itself on entry and clean-up on exit

Option 1 – The player will initialize itself, but will skip the clean-up process

Option 2 – The player will skip both initialization and clean-up

Option 3 – The player will skip initialization, but will clean-up when finished

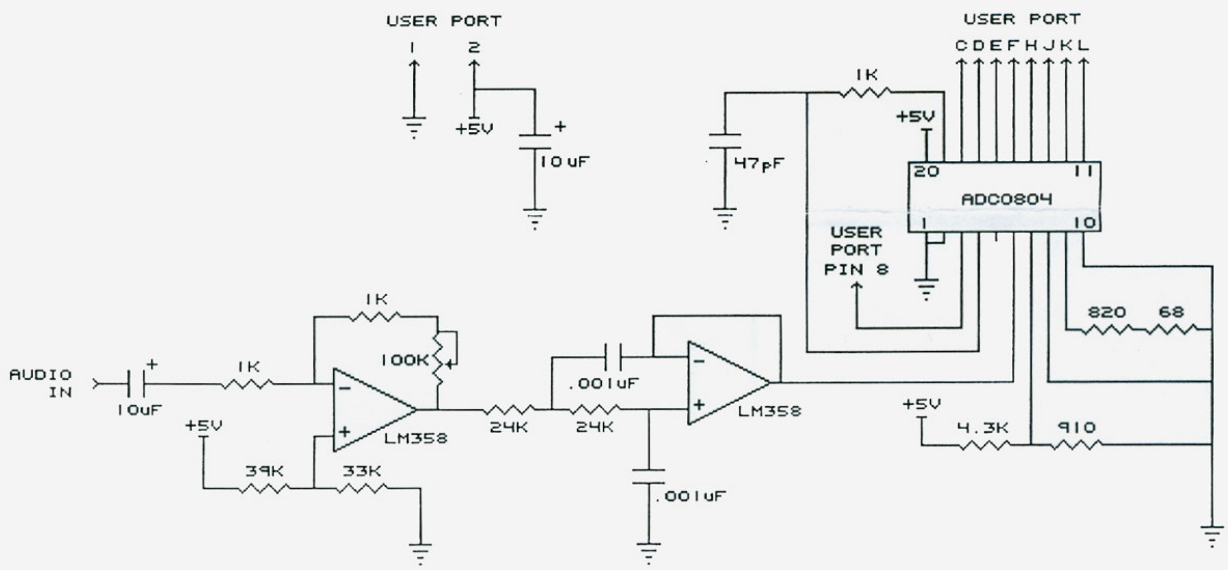
These special options can be used when you want to playa number of samples one after the other without any breaks in between. When chaining samples, you must use option 1 for the first sample, and option 3 for the last sample. You must also remember to set the address of your sample into $fa and $fb before you call the player each time. Note that when using any of the special options the screen will be blanked during playback regardless of the sample rate.

During initialization, the player stores the processor status and memory configuration. These attributes are then restored when the player performs cleanup.

Resource Usage

A,X,Y
$92,$96,$9b,$c3-$c5,$fa-$ff
CIA #2 Timer B
VI C Raster Register

Digimaster Audio Sampler



This schematic is being provided for those of you who have a desire to construct the sampler hardware for Digimaster. You must read the disclaimer in the Digimaster user manual before attempting to build and use the sampler hardware described in this document.

All parts should be readily available from most electronics supply stores. The only one that may be difficult to find is the edge connecter for the user port. The design is fairly straight forward. The audio comes in through an RCA jack, is converted to eight bit digital data, and then fed to the user port through a double sided 12 pin edge connector. For best sound quality, the 10 uF input capacitor should be tantalum.

This sampler will be made available for those who wish to purchase it. Contact Autumn Technologies for pricing and availability.