Studio 16 2.0

AD516 / AD1012

User's Manual

SunRize
INDUSTRIES

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Your AD516 or AD1012 hardware is warranted to the original purchaser for a period of one (1) year from the original purchase date--under normal use and service--against defective materials or workmanship. ANY IMPLIED WARRANTIES ARE ALSO LIMITED TO THE ONE YEAR PERIOD FROM THE ORIGINAL PURCHASE DATE. This warranty is void if the AD516 or AD1012 has been damaged by accident or unreasonable use, neglect, improper service or other causes not arising out of defects in material or workmanship.

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During the above one-year period, defective parts will be repaired, adjusted and/or replaced (at Manufacturer's option) without charge to the purchaser when the AD516 or AD1012 is returned, prepaid and insured, with proof-of-purchase date, to SunRize Industries customer service listed below. UNITS RETURNED WITHOUT PROOF-OF-PURCHASE DATE OR REPAIR FEE WILL NOT BE REPAIRED.

Before returning your AD516 or AD1012 for service, please review the service instructions in this manual.

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FCC Regulations

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

Note: Use of shielded audio cables is required to comply with the class B limits in Subpart J of Part 15 of FCC Rules.

Table of Contents

Introduction

Studio 16 software, when combined with either the AD516 or AD1012 audio card, becomes an advanced hard disk recording system. Both the AD516 and the AD1012 audio card plugs into your Amiga 2000/3000 and adds audio in, audio out, and SMPTE in to your computer. They both include a sound coprocessor (a DSP) as well as high fidelity recording and playback circuitry. The AD516 and AD1012 record audio direct to disk and play back multiple channels direct from disk.

The AD516 is a 16 bit stereo card that suitable for professional Video and CD mastering, the advanced circuitry enables it to playback up to eight channels simultaneously. And the internal SMPTE time code reader allows it to create extensive Cue Lists of sample to trigger to a Video tape. The more economical AD1012 is a 12 bit mono board that includes the same SMPTE support as the 16 bit card and is ideal for video applications. See Appendix B for complete specifications for the AD516 and AD1012.

Included with both the AD516 and AD1012 is the Studio 16 recording and editing software. When combined with an audio card, Studio 16 allows you to record very long sounds directly to your hard disk. Studio 16 allows you to edit and mix sounds.

About this Manual

This manual provides information for installing and using Studio 16. This manual is divided into nine Chapters and a detailed Reference Section. An Appendix and Index is also included.

Chapter 1 .... **Digital Audio** describes the digital sampling process and explains the difference between 8, 12, and 16 bit audio.

Chapter 2 .... **Installation** explains how to install the AD516 and AD1012 in a computer and install Studio 16 on a hard drive.

Chapter 3 .... **Getting Started** includes a brief introduction to sampling, playing, editing, and an overview of common screen elements.

Chapter 4 .... **Hard Disk Drives** provides information on required hard disk specifications, and optional storage devices.

Chapter 5 .... **SMPTE** includes an introduction to the SMPTE Time Code Standard and suggestions for SMPTE applications.

Chapter 1

Introduction to Digital Audio

In recent years, digital audio has become increasingly popular. The most common use of digital audio are currently Compact Discs and Digital Audio Tapes. DATs are replacing analog tapes for professional audio use, just like CDs have replaced LPs for personal use.

Digital vs. Analog

Both CDs and DATs record numbers instead of analog signals. For example, a standard cassette tape records sound by magnetizing a tape. When the sound being recorded gets louder, the tape recorder writes a stronger magnetic field to the tape. When the sound gets softer, the tape player writes a softer magnetic field onto the tape. A DAT recorder, on the other hand, records numbers. For loud sounds, it writes a large number; for soft sounds, it writes a smaller number.

There are several reasons why numbers are better to record than analog signals (such as the magnetic signal on cassette tapes). For starters, it is much simpler to edit digital sound. Using a computer program like Studio 16, it is easy to move parts of sound around or insert one sound into the middle of another. In the analog world this is accomplished by cutting and splicing tape. Another advantage, and the reason the music industry is concerned about DAT, is that when you copy digital audio you always get a perfect copy. Remember that digital sound is just numbers and is manipulated like any other data in your computer. When you copy a disk with your computer, you get an exact copy of the data. On the other hand, every time you copy a magnetic tape, you add a little distortion or "tape hiss".

Converting Analog to Digital

So how do you turn natural analog sound into numbers? You need a sound digitizer like the AD516 or AD1012. The AD516 and AD1012 measure and record the amplitude of a sound. Amplitude is the loudness of a sound signal at an exact moment in time. The process of measuring and recording is referred to as "taking a sample". To digitize a sound, the AD516 (or AD1012) takes a series of samples. It takes a sample, allows a certain amount of time pass, takes another sample, allows the same amount of time pass, takes a sample, etc. As the samples are taken, loud sounds are recorded as larger numbers and quiet sounds are represented by smaller numbers. The amount of time that passes between samples is referred to as the period. See Figure 1-1. Assume the period is 1/44,100 of a second. By inverting the period you can calculate the resulting sampling rate. The inverse of 1/44,100 is 44,100. The sampling rate is 44,100Hz.

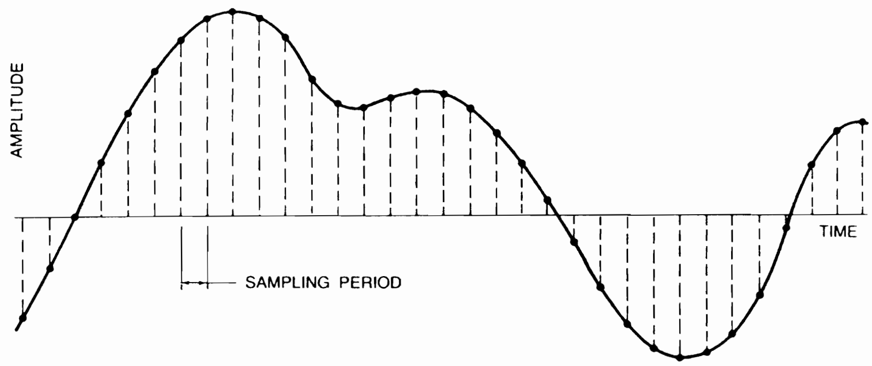


Figure 1-1. Digital Audio Diagram

There is a direct relationship between sampling rate and the maximum frequency you can record. This relationship, called the "Nyquist theorem", states that the maximum frequency you can record is equal to half the sampling rate. So, if you sample at 10,000Hz, all frequencies up to 5,000Hz are recorded accurately. Frequencies over 5,000Hz will introduce distortion into the sound. This distortion is called "aliasing".

Compact Discs always play back at 44,100Hz. This gives CDs a maximum frequency range of 22,000Hz which is outside or on the limits of most people's hearing ability. The AD1012, SunRize's 12 bit audio card can sample up to 80,000Hz, although there is no normal reason to sample this fast.

The second parameter that determines digital sound quality is the maximum sample value. If we let silence be recorded as zero, what value does the loudest possible sound have? This value is the maximum sample value. Sound will oscillate above and below zero by this amount. Using the Amiga's internal 8 bit sound, the maximum sample value is 127. Thus all sounds are recorded as numbers between 127 and -127, with zero being silence. For comparison, the AD1012 has a range of +2,048 to -2,048, Obviously this gives the AD1012 much better sound quality than the Amiga's internal sound. Note that the sixteen bit AD516 has an even larger range of +32,767 to -32,767.

The final thing to notice about digital sound is that it uses quite a bit of memory. Since each AD516 and AD1012 sample takes two bytes, a sampling rate of 44,100 samples per second means that every second of sound is going to take exactly 88,200 bytes. This works out to 5.05 megabytes per minute or 12 seconds per megabyte.

Differences between 8,12 and 16 bit Audio

Digital sound systems generally fall into three different categories: 8, 12, and 16 bit. A small discussion of each class follows. A common reference is given (i.e., "8 bit sounds like an AM radio") as well as the theoretical Signal-to-Noise Ratio (SNR). Keep in mind that the SNR given is the theoretical maximum for a pure sine wave. Practical systems never match this number. However, one aspect of the quality of a real world system is how closely its SNR comes to the theoretical maximum.

8 Bit

The Amiga's internal sound format and SunRize's 'Perfect Sound' are 8 bit. Another common use for 8 bit digital audio is long distance phone calls. Phone calls are often digitized with 8 bits of resolution at the switching station for broadcast via satellite or fiber optic cable. Eight bit audio has a theoretical maximum SNR of -48dBs and its sound quality is often compared to AM Radio.

12 Bit

The AD1012 card is 12 bit. Twelve bit audio is also used in many popular music synthesizers. It has a theoretical maximum SNR of -72dBs. Its quality is comparable to a high quality reel-to-reel tape deck or FM radio.

16 Bit

The AD516 card from SunRize is 16 bit. Compact Disk players and DATs are also 16 bits. Sixteen bit digital audio has a theoretical maximum SNR of -96dBs. Its quality is equivalent to a CD player.

Decibels

In audio systems the common measurement of volume is the Decibel (dB). Studio 16 volumes are also specified in dBs. Decibels are a logarithmic scale used to more accurately represent how the ear hears volume changes. Zero dB means no gain. Attenuation are negative, and amplifications are positive.

For example:

-12dB is equivalent to 25% volume
-6dB is equivalent to 50% volume
0dB is equivalent to 100% volume
+6dB is equivalent to 200% volume

These ratios are determined by the following equation:

dB = -20 log (gain)
e.g. -6 = -20 log (0.5)

Chapter 2

Installation

Hardware Installation

The following installation instructions are the same for the AD516 and the AD1012. Both audio cards plug into a free zorro slot in an Amiga 2000 or 3000. Installing the card is relatively simple. However, if you prefer, your local Amiga dealer can install it for you.

Install the card

1. Turn off your computer.
2. Unplug the mouse, keyboard, monitor and power cables.
3. Remove the 5 screws securing the case. (2 on each side, 1 in the back, top-center)
4. Carefully slide the case off the computer. For more information on removing your Amiga's case, see your Amiga's Manual.
5. Identify the Zorro slots in your computer by referring to your Amiga's Manual. Zorro slots are also referred to as "Amiga 100 pin expansion slots". (Note that any slot in the A2000 or A3000 that the AD516 or AD1012 will fit into is a Zorro slot.)
6. Choose a Zorro slot to receive the card and remove its cover plate (back bracket).
7. Ground yourself by touching your Amiga's metal power supply case. This will remove any static charge built up on your body.
8. Remove the AD516 or AD1012 card from its anti-static bag and plug it into the free Zorro slot.

> Note: When the card is installed properly, only a small amount of the gold connector will show. If the slot has never been used, it may take a lot of effort to push the card in. A gentle rocking motion is usually best.

9. Screw the card's bracket into the Amiga with the screw you removed from the cover plate.

10. Before you replace the case, record your card's serial number on your Registration Form.

11. Replace the case and secure it with the five screws.

Setting Jumpers

AD516 has one jumper. It is indicated by JP1 on the pc board. The AD1012 has two jumpers - indicated by JP1 and JP2. All jumpers are set at the factory and shouldn't need changing.

AD516	JP1 should be set to 6. It selects interrupt 2 or 6.
AD1012	JP1 should be set to 6. It selects interrupt 2 or 6. JP2 should be in place for normal operation.

Connecting Audio and SMPTE

All five jacks are "consumer line level" unbalanced connections. They should be connected to an appropriate source using standard RCA patch cables. For example, connect the left and right "Line Out" on your CD player to the "L - IN" and "R- IN" on the AD516.

Many professional Video Tape Recorders (VTRs) use "balanced" type connections. These will typically have the three pin XLR connectors. If your equipment uses these connectors, you will need to use XLR-to-RCA adapters. These can be purchased at a local electronics store.

The notations on the AD516's and AD1012's brackets follows in figure 2-1.

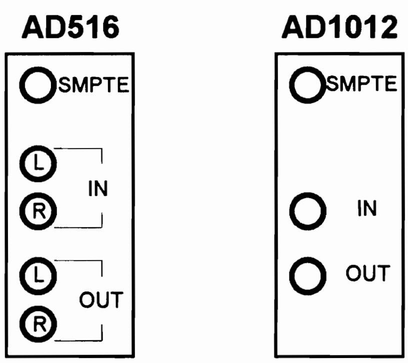


Figure 2-1. Bracket Labels

AD516

The AD516 card has five RCA jacks: SMPTE (LTC) In, Left and Right Audio In, and Left and Right Audio Out.

AD1012

The AD1012, a monophonic card, has the same SMPTE LTC input as the AD516. Its Audio In and Out are standard unbalanced RCA connectors. If you are recording from a stereo source you should record both channels and then later mix them using Transport. Using a "y" connector is not generally recommended.

SMPTE Sources

Although Studio 16 doesn't require an external SMPTE source to run, it is enhanced by the addition. If your deck has a dedicated LTC SMPTE track, simply connect it to the AD516's or AD1012's SMPTE In. If your VTR does not have a dedicated SMPTE track, you can record LTC time code to an unused audio track. Then your SMPTE out will be one of the Audio outs on your deck. For more information on time code, see Chapter 5 -SMPTE.

If your deck uses VITC SMPTE, the SMPTE out on your deck may require a VITC-to-LTC converter. The AD516 and AD1012 do not read "vertical interval time code" (VITC) directly. See Chapter 8 - Third Party Sources for a VITC-to-LTC converter supplier. For information on time code see Chapter 5 - SMPTE.

The AD516 and AD1012 SMPTE reader is designed to handle normal speed variations due to tape motor variance. It is not designed to read time code in "fast forward" modes.

Software Installation

Install Studio 16 on a Hard Drive

1. Turn on your computer and wait for WorkBench to finish loading.

2. Insert the Studio 16 **Disk 1** into a floppy drive.

3. Double click its disk icon.

4. Double click the **Read_Me** icon. The read_me file contains information that was not included in this manual. It may include information related to installation.

5. Double click the **Install.HD** icon.

6. The installation program will run. Figure 2 - 2.

Note: The install program is equipped with on line help and a status line for your convenience.

7. Select the hard drive or partition to put the Studio 16 program on. Your available Amiga DOS devices will appear in the upper left window. Studio 16 must be installed in the root directory of a device, not a sub directory. Click one of the names to select the device. (You do not have to use the same path that you intend to use for your audio files.)

8. Click on the large **Begin Install** button.

9. The installation program will create a directory called 'Studio16' and copy the appropriate files into it. In addition, your s: and libs: directories will be altered.

10. Insert the Studio 16 **Disk 2** when indicated.

11. The installation is complete when an Installation Complete Requester appears.

12. Click **OK** and close the installation window.

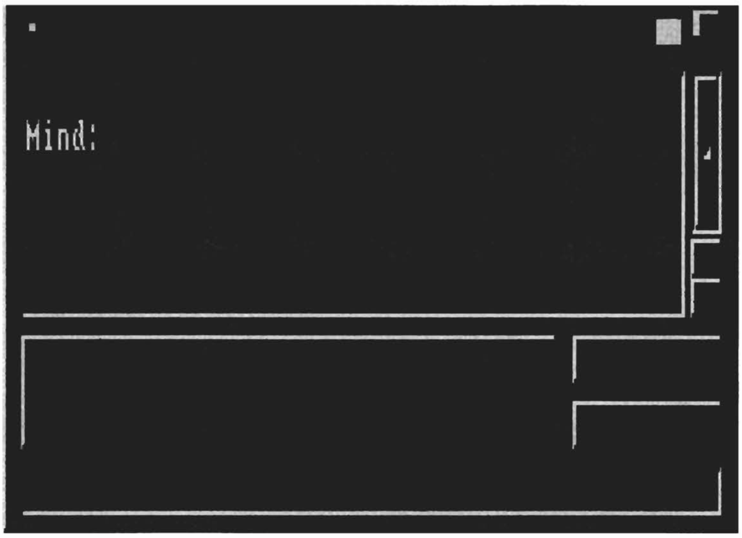


Figure 2-2. Studio 16's Install Program

Load Studio 16

Studio 16 will run under WorkBench 1.3 or 2.0. You can load Studio 16 from the either WorkBench or the Shell.

From WorkBench -

Double click the **Studio16** directory icon.
Double click the **Studio16** program icon.

From the Shell -

Change the current directory to **Studio16**.
Type **`studio16`**.

Running Studio 16 before partitioning your hard drive

Because Studio 16 is very hard disk read/write intensive, it is very important to create a partition on your hard drive dedicated to your working audio files. If you've just received your AD516 or AD1012, and would like to try it out before you backup and partition your hard drive, you can safely run Studio 16 by recording into memory. When Studio 16 is first installed it will set your record path RAM:. This means that all recordings will be made directly to your memory - not to your hard drive. Check the Open List for your record path. It is indicated by a ☒ next to a path name. Make sure RAM: is the selected record path if you have not partitioned your hard drive.

Partitioning your hard disk

Studio 16 can use any Amiga DOS device to record and playback from, (i.e. DF0:, DH0:, RAM:, etc.). And, unless you are recording to RAM:, Studio 16 is very hard disk read/write intensive. Therefore, it is very strongly recommended that you create a separate partition on your hard disk for your audio files you are recording, editing, and playing. If your Amiga crashes while a write is in progress, you may have to reformat your hard disk. If you have partitioned your hard disk and a crash occurs, you will only lose your "working" audio partition. You will not lose anything valuable, i.e. programs and archived samples.

Note: SunRize is not responsible for data lost due to crashed hard disks. Partition your hard drive!

If you are working with a single hard disk, it is a good idea to turn "Reselection" OFF. For instructions on partitioning your hard disk, refer to your hard disk controller manual. If you are using more than one Hard Disk, you should have "Reselection" ON. (Reselection is an option available in your partitioning program.)

When naming your Audio partition. Do **not** name it Studio16. This will confuse the system since there is a Studio 16 assign. Tutorials in this manual will refer to the partition as if it were named "Audio". You may want to name yours "Audio" as well.

Set your Record Path

A Record Path is the assigned location for your new audio recordings. The default path is RAM:. Once you have partitioned your hard disk, you will want to change the record path to a directory on the hard disk. Your record path should be located on your Audio partition.

The following changes the record path from RAM: to a directory on the hard disk.

1. Select Open List from the Applications Menu (^O).

2. The Open List appears with RAM: showing and selected as the record path, note the selection box ☒ next to the path name.

3. To add another path to Open List, select Add New Path from the Open List menu. Select your new path from the requester and click OK.

4. Open List will update and display the two directories. You can continue adding paths (up to eight) to suit your setup. Remove the RAM: path by selecting Remove Path from the Open List menu and selecting it in the requester.

5. Once you have all your paths listed, choose one for the record path by clicking the active box ☒ next to its name. You can change this selection at any time.

6. To save the Open List setup, select Save Preferences from Project menu. This will also save the positions of the windows on the screen. If you don't want Open List showing every time you run Studio 16, close Open List before you SaveSetup.

Studio 16 File Structure

Studio 16 requires the following directories of files to operate. They are automatically copied by the Install program into a Studio16 directory.

Applications
Drivers
Utilities
ProjectMenu

The Studio 16 Install program also places the following files in your libs: directory.

Gadlib.library
Interface.library
Studio.library

Studio 16 should have an assign called "Studio16:" that points to the created Studio 16 directory. The install program automatically creates a file called "s:Studio16Path.config" (an ASCII file that contains the path to the created Studio 16 directory). Unless you issue an assign statement in the Shell or your s:startup-sequence, Studio 16 will issue one the first time it's run based on the "s:Studio16Path.config" file. Therefore, if you move the Studio 16 directory after you have installed Studio 16, you must create a Studio16: assignment that points to the new location.

Chapter 3

Getting Started

This chapter is a brief tutorial on using basic Studio 16 options. Included is description of common screen elements and short tutorials to record, play, and edit a sample. Detailed information on all modules is provided in the Reference Section of this manual.

It is recommend that you open and read the **Read_Me** file on **Disk 1** before loading Studio 16. The file includes updated information that is not in this manual.

Load Studio 16

1. Double click the **Studio16** icon in the **Studio16** drawer. For more detailed instructions see Chapter 1 - Installation.

2. An About window will appear listing the default record path and free disk space. Click **OK** to close it.

Menus and Keyboard Shortcuts

Studio 16 Modules can be launched from the Applications Menu, from the keyboard, or from the Instance List. To select a module from a menu, click the right mouse button and hold it down. Move the mouse pointer over a menu option and release the mouse button to select it.

Keyboard shortcuts are identified in the menu to the right of the option name. Module shortcuts use the control (Ctrl) key, indicated by(^). All other keyboard shortcuts will use the Right Amiga Key(*A*-). To use a keyboard shortcut, hold down either the Control key or the Right Amiga key, and type the letter indicated.

The key to using software efficiently is to memorize the keyboard shortcuts for your most common applications.

Once a Module is launched it may add more menus to the menu bar. For a module's menu options to be available and to use its keyboard shortcuts, you must first make the window active by clicking on it. The title bar of a window will be highlighted when it is active.

NOTE: OK requesters can generally be accepted or rejected with the standard keyboard shortcuts.
For OK, Left *A*-V. For Cancel, Left *A*-B.

Common Screen Elements

Most screen elements are common to the Amiga Intuition standard; however, there are a few gadgets that you may not be familiar with.

Drop List - Indicated by a ↓, allows you access to available choices for an option. Click the ↓ button for a list to appear below the current selection. Use the familiar scroll bar and arrow buttons to locate a new selection. Click on an item to select it.

Active Boxes - Many options can be turned on by activating them. Click the active box next to the option's label, ☒. Deactivate it by clicking the button a second time or by selecting an other option.

Minimize/Maximize - Found in the title bar. Click this button to reduce a window to its smallest possible dimensions. Click the button again to enlarge the window to its previous dimensions and location.

Depth - Located in the top right corner of all windows, the Depth button allows you to bring a window to the front or move it to the back. Click the button once to move it to the front. If the window is already in front, the single click will move it to the back or underneath all other windows. Also selecting a module from the Applications Menu will bring it to the front. If ClicktoFront is selected in Preferences, this button will not appear. Windows are then brought to the front by clicking anywhere on them.

Resize - In the lower right side of most windows is a small triangle. Click and drag this triangle to resize a window. As windows are reduced in size, text in the window and title bar will be truncated.

Record and Play

Once you have installed and loaded Studio 16 you are almost ready to record.

Before you can record, you need to connect your audio source. Any audio source with a "Line Out" may be used, preferably a CD player. Use a short RCA patch cable to hook up the CD's Line Out to the board's Audio IN jack(s). If your audio input is from a microphone, you will need an external device, like a mixer, to turn the mic output into an RCA Line Out. The card's Audio OUT jack(s) can be hooked up to a receiver or an amplifier system's CD/AUX Line In. If you have a LTC SMPTE time code source, plug it into the top jack: SMPTE IN.

Now that you are ready to record, there are two methods of recording. One uses the Recorder module and the other records from Transport.

Record & Play a Sound - METHOD 1, using Recorder and Open List

Record with Recorder

1. Load Recorder, Open List and Meters from the Applications Menu. Close any other modules that are open. Figure 3-1.

2. Check your record path in Open List. If you have not partitioned your hard drive, it should be set to RAM:. (The record path is identified by ☒.)

3. Turn on your audio source.

4. Click **Monitor** in the Recorder window. You should hear the audio playing through your output device.

5. **AD516 ONLY** - To make a stereo recording, select the L and R channel from the Recorder Menu so that both channels have a ✓ next to them. Make sure you have audio connected to both the AD516's audio inputs.

6. Activate Meters clicking by on the window. From the Channels Menu, select the Input, Output and Play 1. These are easily selecting by holding down the right mouse button, and clicking the channels with the left mouse button. You will now see activity on the input and output meters.

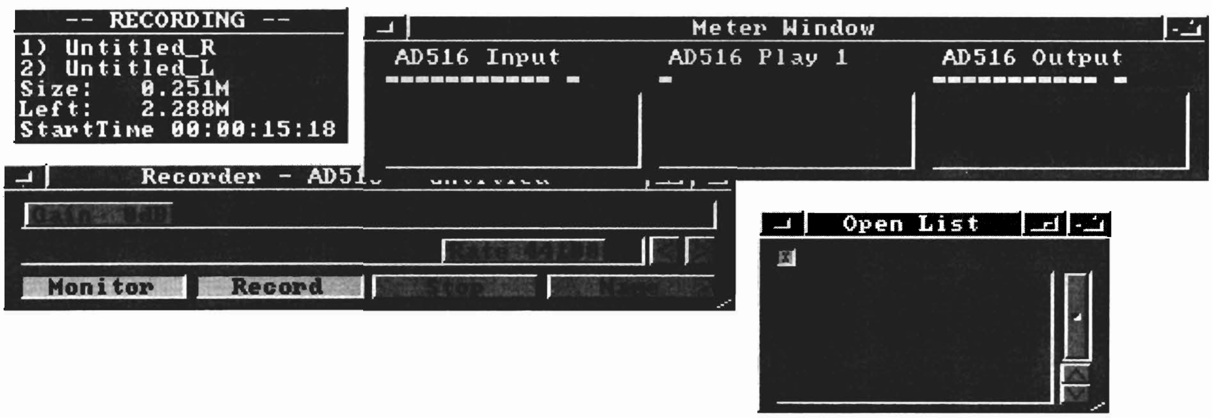


Figure 3-1. Recorder and Meters

7. Set the proper **Gain** setting by watching the meters for clipping. Clipping results from the gain being set too high. It is indicated on the meters by the VU meter going into the (+) section, the LED meter flashing on the far right side, and the scrolling graph meter peaking off the top and bottom. Adjust the **Gain** so that clipping rarely, if ever, occurs. Figure 3-1. See the Meter Reference Section for more information.

8. **AD516 ONLY** - Stereo samples will be shown in the scrolling meter by channel. The input from the left channel will be graphed on the top of the graph, and the input from the right channel will be graphed on the bottom of the graph. Disconnect one of the inputs to see the effect on the scrolling graph.

9. **AD1012 ONLY** - AD1012 owners have the added option of setting a variable lowpass filter. Activate **Auto Filter** to set an appropriate cutoff filter for your sampling rate. (The AD516 has a digital filter. It's automatically activated whenever you record.)

10. Select a sampling **Rate**. Keep in mind that 48KHz is the standard rate for DAT players. Anything more than this is wasting memory. For video purposes you may want to record at about 30,000Hz. This gives a 15KHz frequency response, more than adequate for narration and sound effects. (The nearest available sampling rate for the AD516 is 32,000Hz, the AD1012 can record at 30,120Hz.)

11. Click the **Name** button and type in a name for your sample. The default is Untitled.

> Note: You can name the sound before you start recording, or you can rename the sample in the Open List later.

12. To begin recording, click **Record**. A window will appear in the middle of the screen indicating that recording is in progress, Figure 3-2. Also displayed is the name(s), the size of the sample recorded so far, and the remaining disk space. These numbers will change as the record proceeds. And, unless you're recording to RAM, your hard disk light will flash.

Figure 3-2. Recorder's Progress Indicator

If you have SMPTE time code plugged into the AD516 or AD1012, the recorder will show the time your recording began. Later, if this sample is loaded into the Editor or the Cue List, this SMPTE time code will enter itself with the sample.

13. To stop recording, click the **Stop** button. If you continue to hear sound, it is the monitoring of the audio source. Click the **Monitor** button to stop monitoring. The "monitor" and "record" functions are independent. Also, your hard disk light may continue to flash for a few seconds after selecting stop. This is normal.

Play from Open List

There are a few ways to play back a sample. To play back from Open List:

1. Launch Open List from the Applications Menu, (^ O). Figure 3-3. (You can close any other modules that are open.)

2. Select the sample you recorded earlier by clicking on its name. It will be in the active directory. The active directory, or record path, is indicated by ☒.

3. **AD516 ONLY** - To playback a stereo sample, you need to select both the samples that were recorded, (e.g. Untitled_L and Untitled_R). Select both samples by shift-clicking. (Holding down the shift key while clicking on multiple sample names.)

4. Play the sample(s) by selecting **Play Sample** from the Open List menu or by typing *A*-P.

5. To stop the playback early, select **Stop Playback,** *A*-S.

Note: When playing back multiple samples, all samples must have the same sampling rate.

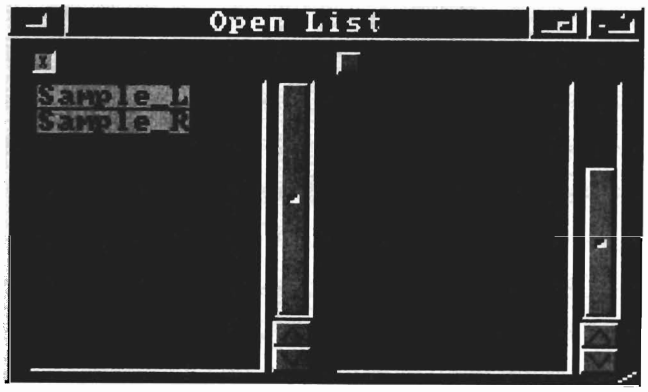


Figure 3-3. Open List

Record a Sound - METHOD 2, using Transport

You can also record and playback from within the Transport module. The benefits to this method are that you can record the output of the card allowing you to mix sounds together, and you can play sounds while recording. The following tutorial makes a simple recording from the input and then plays it back.

Record from Transport

1. Verify your Record Path in Open List. If you have not partitioned your hard drive, the record path should be set to RAM:, (indicated by ☒).

2. Launch Transport and Mixer from the Applications Menu. Close all other modules that may be open. Figure 3-4.

3. Turn on your audio source.

4. Monitor your input by raising the input level on the Mixer. You should hear the audio playing.

5. In Transport, the sampling rate and gain level are set from Handler in the Options Menu. Similar to Method 1, set an appropriate **Gain**, sampling **Rate**. AD1012 users should also select a **Filter** setting. Then click **Exit**. You may want to load Meters to adjust the **Gain**.

6. In Transport, activate a record track (☒) and type in a name for the recording, the default name is Untitled.

7. **AD516 ONLY** - To make a stereo recording, activate both record tracks (☒) and enter names for both channels, (e.g., Untitled_L and Untitled_R).

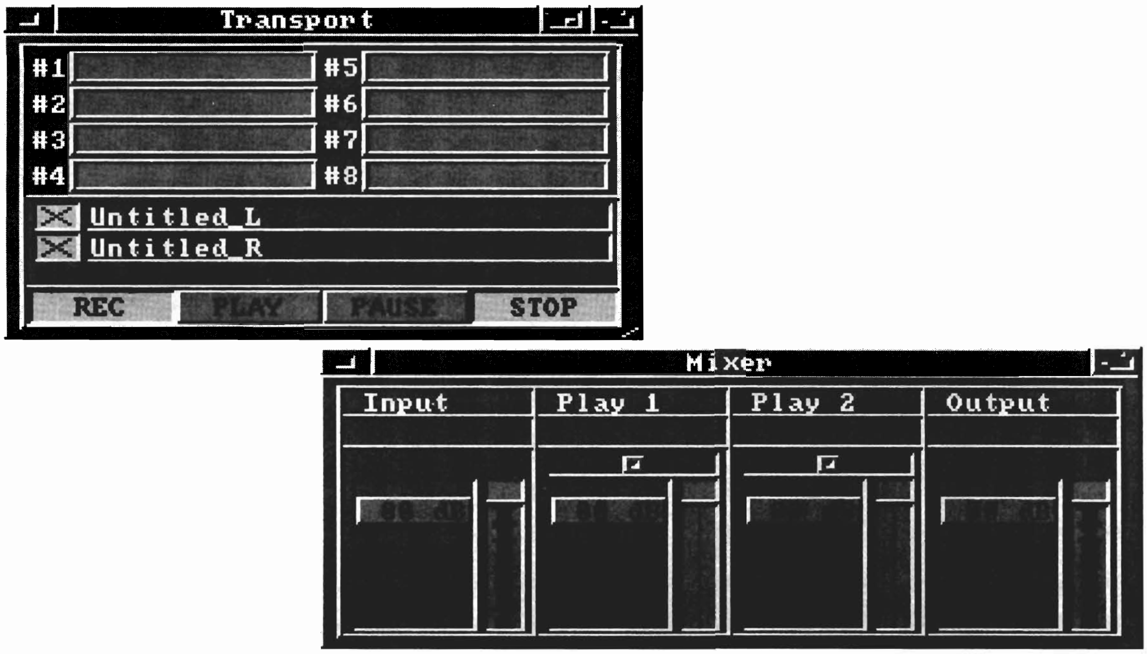


Figure 3-4. Transport and Mixer Modules

8. Click **Record** then **Play**. Both buttons must be depressed to begin recording.

9. Click **Stop** to end the recording.

10. Stop monitoring by decreasing the input volume in the Mixer to -∞. (-00dBs)

11. Your new sample(s) will now be listed in Open List.

Play from Transport

1. Launch Open List from the Applications Menu, (^ O).

2. Drag your sample(s) from Open List into Transport and drop it in a playback track at the top of the window.

3. **AD516 ONLY** - To playback a stereo sample, both samples must be played simultaneously. So, drag both samples, (e.g. Untitled _L and Untitled_R) into the Transport playback tracks.

4. Select Play in Transport (*A*-P).

Edit a Sample

To make changes to a sample, it must be loaded in an editor. This part of the tutorial covers basic navigation within the editor and use of the non-destructive cut, copy and paste.

Load a Sample Editor

1. Load Open List from the Applications Menu (^ O).

2. Select the sample you want to edit and select **Edit Sample** from the Open List Menu or type *A*-E.

3. **AD516 ONLY** - To edit a stereo sample, select two samples in the Open List (ending in _L and _R) by shift-clicking and then select Edit Sample (*A*-E)

4. An Edit window will appear with your sample(s) graphed. Figure 3-5.

5. Select **Play All** from the Editor Menu to hear the sample (*A*-P). The keyboard shortcut *A*-P is one the most useful short cuts, it causes samples to play in many of the modules.

Copy a Range and Duplicate it within the Sample

1. Mark a range over the area of the graph you want to duplicate. To mark a range, click on the graph and drag the pointer to the left or right. To alter the size of an existing range, click on one edge of the range and drag it to a new location.

2. Select **Play Range** from the Editor Menu (*A*-L) to hear the marked range. Figure 3-5.

Figure 3-5. Edit Window

3. Select Non-Destructive Copy from the Edit Menu.

4. Click on the graph to select an insertion point for the copied range.

5. Select Non-Destructive Paste, and Insert at Start, (*A*-V). (Start refers the beginning of the marked range. When you just click on the graph, you create a very small range.)

6. Click **Play All** to hear the new version of the sample. **Undo** will return the sample to its previous state.

> Note: Because the previous copy and paste were Non-Destructive edits, **Undo** is able to cancel the change. Destructive edits are <u>not</u> undoable. Having a choice between destructive and non-destructive editing allows you to have more control over your samples. Destructive editing actually changes the data on your hard drive as you alter a sample in the editor. However, Non-destructive edits do not destroy any data on your hard drive. It just remembers which edits you want to perform on the sample. Non-destructive edits can be made permanent and free hard disk space by selecting Make Permanent. See the Reference Section on the Editor for more information.

Delete a Range

1. Mark a range to delete using the previous technique.

2. Select Non Destructive Delete from the Edit Menu, (*A*-D). Since the delete is non-destructive, it will not remove data from your hard disk. To free hard disk space, use the destructive delete or select Make Permanent after doing the non-destructive delete.

Create a Region

Regions are named ranges within samples. Once created in the Editor, they can be dragged and dropped into the Cue List as an entry, or into Transport for playback.

1. In the Editor, select a range to be named as a region. For details on marking ranges refer to the technique mentioned previously in this section.

2. From the Options Menu, activate Show Regions. An Editor Regions List will appear, Figure 3-6.

3. When the Editor Regions List is active, a Region Menu is available in the Menu bar.

4. Select Add Region from the Region Menu to create a region from the marked range in the Editor.

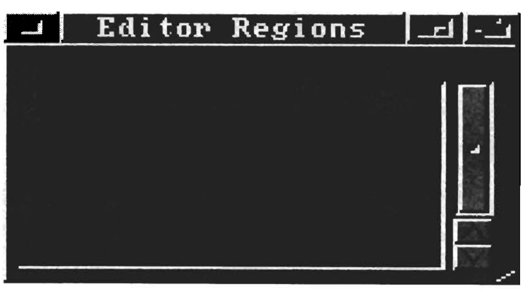


Figure 3-6. Editor Regions List

5. When prompted, enter a name for the region.

6. You can now drag this region into Transport or Cue List for playback. (Regions can also be dragged from the Open List when Show Regions is activated from the Open List menu.)

Convert a Sample

When a sample is recorded it is usually stored in the Studio16_2.0 file format. This default format is selected in Preferences from the Project Menu. Studio16_2.0, like AIFF, is a 16 bit format. It includes special data that keeps track of non-destructive edits and marked regions. To export a sample to another audio program you will probably have to convert to another file format. Studio 16 will convert to: AIFF, IFF 8SVX, RAW, CDTV RAW. Below are the steps to convert a sample to an IFF-8SVX format.

1. Load Open List from the Applications Menu.

2. Select a Sample to Convert.

3. Select Convert from the Open List Menu.

4. Select **IFF-8SVX** for the file format. Figure 3-7.

5. Click **OK**.

6. A Save File Requester will appear. Enter a directory and file name for the new 8 bit sample.

7. Click **OK**.

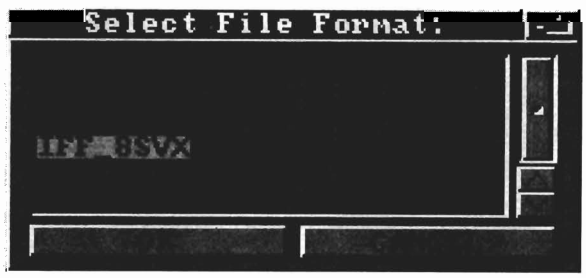


Figure 3-7. Convert Sample

8. You can still load this new file into Studio 16, but because converting a 16 bit sample to an 8 bit format drops bits, the sample won't sound as good as it did before the conversion.

9. The original 16 bit sample is still on disk. You can delete it from Open List.

Note: See the Reference Section on Open List for a description of the available file formats and more detailed instructions on converting.

Converting files to IFF 8SVX can be especially useful if you need to conserve disk space and you're working with samples that don't require a high SNR. Sound effects like explosions will likely sound the same whether they're 8 bit or 16 bit files. Both 8 and 16 bit files can play simultaneously, if they have the same sampling rate. Plus, the 8 bit file will only take up half of the disk space it did when it was a 16 bit file. (Note that samples recorded with the 12 bit AD1012 are stored in a 16 bit format.)

Quit Studio 16

To quit Studio 16, select **Quit** from the Project Menu. Before clicking OK, you should check the Open List for any samples that are in the RAM: directory. These samples should be moved to a hard disk before turning off your Amiga; otherwise, they will be lost.

To save the position of all the open windows and their settings select Save Setup from the Project Menu before selecting **Quit**.

Shell Commands

There are six shell commands included with Studio 16. These commands allow you to access Studio 16 samples and SMPTE without actually loading the program.

StudioClose
StudioOpen
StudioPlay
StudioQuery
StudioStop
StudioWait

See the Reference Section for more details on a particular command.

Chapter 4

Hard Disk Drives

Studio 16 usually uses a hard disk to store audio files. Although it is possible to use RAM for sample storage (add RAM: path to Open List), this is not common. Hard disks are preferred because they can store large amounts of data relatively cheaply.

About Hard Drive Specifications

Hard disks are primarily rated by their "average seek time". This is the average time it takes the read/write head to move between two random tracks. For example, the hard drive in an Amiga 3000 is often an "11 ms, 105 Meg Quantum." This translates to a drive that holds 105 megabytes of data, is manufactured by Quantum, and has an average seek time on reads of 11 milliseconds (which is quite fast). This is a little deceiving because the Quantum drive has a track buffer that makes reads appear faster than writes. The average write seek time on the same drive is 19 ms.

A second parameter of drives is the "data transfer rate" from the drive once the seek is completed. The hard disk controller you use can affect this as much as the drive itself. In general, slow data transfer rates are only important on optical drives. For example, a fast drive with an A3000 hard disk controller may run at 1.9 MB per second. However, a typical Magneto-optical drive runs at 100K per second for writes, and 300K per second for reads.

How fast does my hard drive need to be?

Studio 16 will work with almost any drive you can buy these days. However, the faster the better. For example, the A3000 105 MB Quantum with an average seek time on reads of 11ms and 19 ms on writes is fast enough for about 6 tracks at a 44.1 KHz sampling rate using the AD516.

But even slower drives will work. For example, the "SyQuest" removable media drive has an average seek time of 26ms. It is possible to do multiple tracks with this drive. However, you may have to reduce your sampling rate or the number of tracks you are playing back. Note that the AD516 is much more efficient than the AD1012. As a general rule you can play more tracks at faster rates with the AD516.

Removable Media Drives

SyQuest is a popular removable media disk drive. In testing, we found that you can play multiple tracks off a SyQuest drive, although you may have to reduce the sampling rate or the number of simultaneous tracks. You may also need to increase the "Channel Buffer" in Preferences.

Optical Drives

Although we haven't tested it in house, we have reports from AD1012 owners recording and playing one channel of audio with a Ricoh 600MB magneto-optical drive. This drive has an average seek time of 33.7 ms with read and write rates of 300K/second and 100K/second respectively. In order to work, the "Channel Buffers" in Preferences were increased to 1MB. Note that performance will be improved with the AD516.

Another user reports to playback 2 tracks with a Maxtor Tahiti II - 1 GB magneto-optical drive using the AD1012 in an A3000. The access time for this drive is 90-120 ms.

For more information on the above drives, contact the manufacturers, listed in Chapter 8.

Hard Drive Space Requirements

Memory usage is entirely based on the sampling rate of a sample. The following chart provides a Meg per Minute guide for common sampling rates when recording 16 bits. The formula used to derive the chart is: $MB = \text{Sampling Rate} \times 2 \times 60\ \text{sec/min} \div 1024\text{byte/k} \div 1024\text{k/MB}$. Or, more simply, 1 second of sound = $2 \times$ Sampling Rate.

When recording in stereo, double the space requirements listed below.

Sampling Rate	Space for 1 minute (1 track) (60 seconds)	Space for 30 minutes (1 track) (1800 seconds)
11,000	1.3 MB	37.8 MB
22,000	2.5 MB	75.5 MB
32,000	3.7 MB	109.8 MB
44,100	5.0 MB	151.4 MB
50,000	5.7 MB	171.6 MB

Fragmented Hard Drives

As drives are used, they tend to become cluttered. The operating system starts to spread files out across tracks that are not close to each other. As a result, it takes longer to read the files because the hard drive must seek farther between tracks. This is called "fragmentation" and becomes a particular problem as the hard disk becomes full. If your hard disk or audio partition is fragmented, you may have trouble playing sounds. You can purchase programs that will "optimize" or "de-fragment" your hard disk. They scan your hard disk and rearrange the data so that all the data for a particular file is near each other.

Disk Errors

Hard drives aren't fool proof, and it is almost inevitable that you will at some point encounter a disk error. This can manifest itself as a "Read" or "Write" error. Or, occasionally as a "Can't Validate Drive" error. Sometimes clicking "retry" or "cancel" on the DOS requester presenting the error will cause the error to disappear. However, if this happens you should start to worry because the error will probably reappear later, and can mean a failing hard drive.

Of course your best advice is to backup your hard drives. They can be backed up to a SCSI tape drive or a SCSI optical drive. However, due to the time and expense involved, many people don't bother. Just be warned that a hard disk "crash" is not just a possibility, it is very likely that it will happen to you eventually. For more information on preventing disk errors, and recovering data after a disk error, refer to Chapter 7 - Troubleshooting.

If your hard drive crashes, and you're not sure what to do, take your Amiga to your Amiga dealer. Their service department should be experienced in attempting to restore data from corrupt hard drives.

Chapter 5

SMPTE

Introduction to SMPTE Time Code

SMPTE (Society of Motion Pictures Television Engineers) time code is a standard way to keep track of time or position on tape. SMPTE (pronounced "simp-tee") time code is most common in video production, but is also used in film and music production. In general, SMPTE time code specifies position and timing information in terms of frames. The format is:

HH:MM:SS:FF (HOURS:MINUTES:SECONDS:FRAMES)

(01:12:06:02 refers to 1 hour, 12 minutes, 6 seconds, and 2 frames)

Time code is used to synchronize events. With Studio 16 you can specify that a sound will trigger at a specific time code. For example, you may find that a door begins to open at exactly 1 minute, 2 seconds, and 15 frames into your tape. You can set up Studio 16 to trigger a "door open" sound effect at 00:01:02:15 with the Cue List.

SMPTE time code is used to synchronize events in music production as well as video production. For example, a musician with a multi-track tape recorder may want to synchronize it with a MIDI sequencer. This can be accomplished by "striping" time code onto the multi-track, and then running the time code out of the multi-track, and into the computer. The computer can then run a MIDI sequencer capable of "slaving" to the tape by following the time code. Studio 16 and the Bars&Pipes sequencer allow you to accomplish this without an external multi-track. Studio 16 can act as a digital multi-track tape deck, and it will sync to Bars&Pipes through "internal" time code.

VITC or LTC SMPTE

There are two basic types of SMPTE time code: LTC (Longitudinal Time Code) and VITC (Vertical Interval Time Code, pronounced "vit-see"). LTC is written on the audio track of the tape deck, and VITC is embedded into the video signal. VITC has the advantage of being available constantly, even when the video deck is paused. Whereas, LTC requires that the deck be playing for time code to be available. However, LTC time code is less expensive, and works on any audio tape recorder--not just video decks.

Both LTC and VITC store the same basic information--the frame number that specifies where the tape decks record/play heads are currently located.

Both the AD516 and AD1012 include a LTC SMPTE reader. If your deck outputs VITC, there are translators available that will convert the time to LTC which is read by the AD516 or AD1012. See Chapter 8 for a supplier.

Frame Rate

One final aspect that needs to be specified when selecting a time code format is the frame rate, or number of frames per second. SMPTE supports the following:

24 fps	Motion Pictures
25 fps	European Video
29.97 fps "non-drop frame"	USA color video
30 fps "non-drop frame"	USA B&W video or music
30 fps "drop frame"	USA color video

When SMPTE was first introduced in 1967, it was for 30 frames per second used in black and white TV. However, when NTSC color TV was introduced, it used the slightly slower rate of 29.97 frames per second. This is approximated in "drop frame" SMPTE by dropping 108 frames every hour. Drop frame is used so that "real time" or "clock time" will match the time code marked on the tape. However, drop frame can cause problems because of the missing frames. So, 29.97 non-drop frame is often used.

Striping Time Code

In order to use time code, you must first "stripe" your tape. That is, you record time code onto the entire tape area you are planning to use. Depending on the camera or video deck you use, this may be automatic or manual. Or, you may have a dedicated "Time Code Out" jack on your deck.

Studio 16's internal SMPTE generator does not output time code for striping.

To stripe LTC onto your tape, you will need an LTC generator. Time code generators can be part of a video cameras or a video deck. Or, they can be dedicated generating boxes, or software programs, namely the SunRize SMPTE Output module. Chapter 8 lists manfacturers of LTC generators.

Contact your dealer or SunRize for more information on SunRize's SMPTE Output module that outputs LTC time code for striping tapes.

SMPTE for Video Applications

Time code is essential in any kind of professional video production. It allows precise specification of video and audio edits. With Studio 16, you use SMPTE time code in conjunction with the Cue List to synchronize audio effects, music, and narration with the video tape. These audio clips reside on your hard disk until you create the final master tape. During your edit sessions, the Cue List will track your tape's time code and trigger sound effects as specified. You will be able to listen to the audio coming off the computer as you watch the video.

SMPTE for Music Applications

SMPTE time code was originally designed for video applications but with the success of MIDI (Musical Instrument Digital Interface) it became apparent that it would be useful to use SMPTE time code to time

MIDI sequencers and to communicate with other MIDI devices. MIDI quickly adopted a MIDI Time Code format. MIDI Time Code is easily converted to SMPTE Time Code and vice versa. To drive external MIDI equipment you may need a SMPTE to MIDI time code converter. Chapter 8 lists a source for such a device. MIDI time code is used as a timing source for sequencers, to sync two tape decks together or to sync a tape deck with a MIDI sequencer among other things. When Studio 16 is used along with Bars&Pipes Professional, you get complete sequencing capabilities as well as hard disk recording. Your MIDI sequences can have vocal tracks and guitar solos without having to use an expensive multi-track tape deck that tracks MIDI time code. See Appendix C: Bars & Pipes for details.

Chapter 6

Cue List Tutorial

The Cue List module allows you to trigger sounds at specific SMPTE time codes. The Cue List itself can contain any number of entries. Each entry specifies a sample name and a time code indicating when the sample will play. In addition, you can enter a remark for each sample. The Cue List allows you to trigger up to eight sounds simultaneously on the AD516, and four sounds on the AD1012. (The actual number of sounds you can play simultaneously depends on your Amiga's CPU speed and hard disk speed.)

The following Cue List tutorial is created by loading the demo sounds from the Studio16:Samples directory into the Cue List Module along with SMPTE Start Times. The samples are then triggered by the Internal SMPTE Generator.

The final outcome is the sound of a doorbell being rung a few times, then a door knock, followed by a door opening and closing. Keep in mind that the demo samples were sampled at 8 bits with a very low sampling rate in order to keep their file size small enough to fit on the floppy disk. Your samples will be sampled with more bits and at a higher sampling rate. They will sound much better.

Add Studio16:Samples path to Open List

Before triggering any samples from the Cue List, you must make sure the samples are in the Open List. This tutorial uses samples that are included in the Studio16:Samples directory.

1. Load Open List from the Applications Menu, (^ O).

2. Select Add New Path from the Open List Menu (*A*-A).

3. Select **(DIR)Studio16** in the device list, and then select, then **(DIR)Samples**. Click OK.

4. Open List will update with Samples as a new directory containing the following files.

   CloseDoor **DoorBell** **DoorKnock** **OpenDoor**

5. To play the individual samples, you can click on the name and select play sample from the Open List Menu, or type *A*-P.

Enter Cue List Entries

Entering a Cue List can be done in two different ways. The first method is detailed below. Enter the items by dragging sample names from the Open List or Editor Regions List. Entries can later be edited in the edit fields at the bottom of the Cue List window. The alternative is to load your word processor and type a list then load it into Cue List. See the Cue List Reference Section for formatting details.

SHORTCUT - The following Cue List has already been created and can be loaded by selecting Load from the Cue List Menu. The default directory is Studio16:CueLists. Select **Tutorial.List** and click OK. Once the List is loaded, you can skip the following steps and go to the next section, "Trigger the Cue List".

Entering the Tutorial Cue List

1. Load the Cue List from the Applications Menu. Open List should already be open with following samples listed in the RAM: directory.

   CloseDoor **DoorBell** **DoorKnock** **OpenDoor**

   Click on **DoorBell** and drag it into the Cue List. Notice the SMPTE start and stop time will automatically update. Cue List starts the first sample at 00:00:01:00 (1 second). The end time is calculated from the length of the sample.

> Note: To Play **DoorBell** , highlight the sample in the Cue List and select **Play Sample** from the Cue List Menu or type *A*-P. This is just like playing the sample from Open List. To play the samples in order, you have to activate a SMPTE source. This is explained in the next section.

3. Next, drag **DoorBell** into the Cue List again. And then a third time. So you have three entries for DoorBell. The Cue List will automatically set the start times of each sample to trigger the samples 16 frames behind one another.

4. Now drag Door Knock, Open Door and Close Door into the Cue List. You should now have six entries in your Cue List.

5. If you dragged the samples in a different order, you may have to shuffle them around to have them look like Figure 6-1. Change the start time of an entry, by typing in the Edit Line, will causes the Cue List to resort by start times. You can also use the menu commands Add and Delete Entry to reorganize your list.

6. Change a start time - Highlight an entry in the list. Delete the existing start time in the edit line at the bottom of the Cue List window. Enter in a new time, and hit return. You do not have to type colons ":".

7. Replace an entry - You can also drag the sample in again from the Open List and drop it in the edit line along the bottom of the Cue List to replace the selected entry.

8. Once the Cue List looks like Figure 6-1, you are ready to trigger.

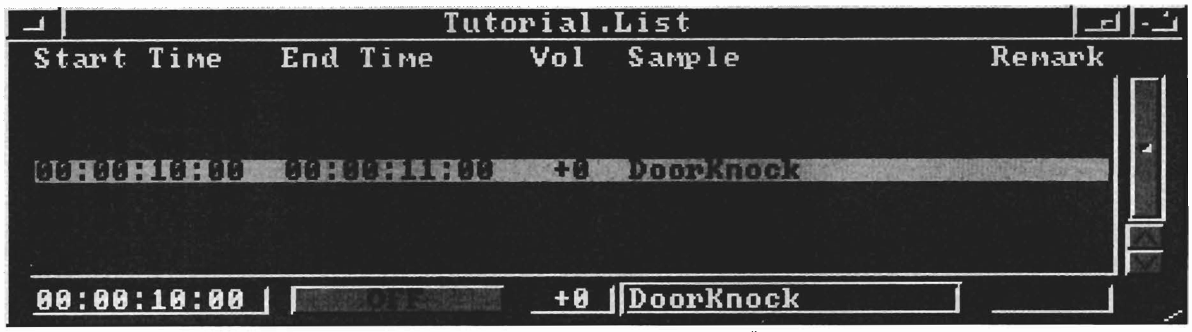


Figure 6-1. Tutorial. List

Note: In the preceding tutorial, the Volumes of the samples were not altered. To lower or increase the volume of a sample, change the entry in the **Vol** field. You can also, enter an optional remark for each sample. Select an entry and type a comment in the Remark field. Always hit the Return key, after you alter a field. This updates the list.

AD516 ONLY: Because the sound effects used in this tutorial were recorded in mono, they playback on the right channel. To make the samples playback on both channels, you will need to enter a pan statement in the remark field. See figure 6-2 for the correct format. The Cue List reference section has more information on pan.

Trigger the Cue List

METHOD 1 - Internal SMPTE Generator

1. Load the SMPTE Generator.
2. Turn the Cue List on, by clicking the **ON/OFF** button.
3. Trigger the Cue List by clicking **Play** in the SMPTE Generator module.
4. Each sample will trigger at the appropriate time. The entry in the Cue List will be highlighted as it is playing.
5. After all the samples have been triggered, click **Stop** and then **Return** in the SMPTE Generator module to reset. The reset time should be 00:00:00:00.

NOTE: To stop the triggering after it has started, simply turn the Cue List **OFF**.

METHOD 2 - External SMPTE Generator

External time code will generally come from a professional VTR tape striped with SMPTE time code, or an external SMPTE time code generator. Your external SMPTE time code source must be connected to the AD516 or AD1012 card. The top RCA jack on the card is the SMPTE IN jack. Use an appropriate cable to attach your external SMPTE source to the card.

To tell Studio 16 to listen to your external time code you must:

1. Close Studio 16's SMPTE generator if it's open.

2. Turn the Cue List on, by clicking the **ON/OFF** button.

3. You can then trigger the Cue List by activating your external SMPTE source.

> Note: It may be useful to open the SMPTE Monitor to display the SMPTE time code coming into the AD516 or AD1012. When you start your external time code source, the SMPTE monitor will display the incoming time code, and your Cue List will trigger.

Save a Cue List

1. To save your Cue List as a Cue List file, select Save As from the Cue List Menu. A Save Cue List File Requester will appear with Studio16:CueList as the default directory. This directory is reserved for your Cue List Files.

2. Enter a name for the List and click **OK**, or hit return.

> Note: Saving a Cue List saves a text files with the list of entries only. It does not save the samples that are in Open List.

Delete a Cue List

You can delete a Cue List File from WorkBench 2.0 or from SHELL in WorkBench 1.3 or 2.0. To delete an entry with a SHELL:

Open a SHELL from WorkBench, and typing the following.
(***name*** should be replaced with your file's name)

```
cd Studio16:CueLists↵
dir↵
delete name.cue↵
endcli↵
```

Chapter 7

Troubleshooting

The following symptoms and their solutions are covered in this chapter.

I. Can't edit a sample in the Editor
II. Can't hear input
III. Can't hear playback
IV. Can't select correct sampling rate in Editor or Transport
V. Card communication errors or GetW/SendW errors
VI. Cue List won't trigger
VII. Flashing screens or skipping/repeating sound
VIII. Full hard disks
IX. Gain won't adjust
X. Graph doesn't match sound
XI. Hard Disk Read/Write errors
XII. Modules won't open
XIII. Playback is out of sync in Cue List or Transport

If the following recommendations do not solve your problem, call SunRize technical support for more information.

I. Can't load a sample in the Editor

a. **Problem** - The sample is being used by another module, usually Cue List.
Solution - Simply turn OFF the Cue List by clicking the ON/OFF button at the bottom of the window, or close the Cue List window.

II. Not being able to hear input

a. **Problem** - Mixer level is set too low.
Solution - Open the Mixer from the Applications Menu, and set the Input and Output Channels' volume level to +00dB.

b. **Problem** - Input cables are not connected correctly. (You know this is the problem if you don't see Meter activity on the Input channels.)

Solution - The audio In jack(s) must be connected with an RCA patch cable to the line out of an audio source. Make sure the audio source is turned on.

c. **Problem** - Gain chip is malfunctioning -- only a possibility on the AD1012.
Solution - If you do not see activity on the input channel of the Mixer, and you're sure your audio source is connected properly, the Digital Pot on the Gain circuit may be bad. Especially if you can barely hear the playback, or changing the gain level has no effect on the volume. Call SunRize technical support if you suspect this problem.

III. Not being able to hear playback

a. **Problem** - Mixer level is set too low.
Solution - Open the Mixer from the Applications Menu, and set the Input and Output Channels' volume level to +00dB.

b. **Problem** - Output cables are not connected correctly. (You know this is the problem if you can see Meter activity on the Output channel.)
Solution - The audio Out jack(s) must be connected with an RCA patch cable to a mixer or the CD/AUX input of a receiver or amplifier. Make sure the receiver is set to CD input and is turned on, also check the speaker connections.

IV. Can't select the correct sampling rate in Editor or Transport

a. **Problem** - The Set Sample Parms or Handler display will not show all the rates available by dragging the selector knob.
Solution -Use the arrow buttons next to the slider or click in the slider to the left or right of the knob to see all the available increments.

a. **Problem** - Sample was recorded with a different type card.
Solution -If the sample was recorded with the AD1012 the same sampling rates will not be available for the AD516, and vice versa. Use the resample option in the Editor to change sampling rates or re-record the sample.

V. Card Communication Errors (e.g. "Comm Error #2", "GetW", "SendW" or "DSP Init. Failed")

a. **Problem** - This very rare error usually means a hardware problem-either with your Amiga or the card.
Solution - Try resetting your Amiga, or turning it off and then back on. If this doesn't help, try re-seating the card in another expansion slot. If this still doesn't solve the problem, call SunRize technical support for more information.

VI. Cue List won't Trigger

a. **Problem** - Most often, the Cue List is just turned **OFF**.
Solution - For the Cue List to listen for time code and preload samples, it must be turned on. Simply click the ON/OFF button at the bottom of the window to turn Cue List **ON**.

b. **Problem** - Entries are not in the Open List, or they contain a typo.
Solution - When a sample is entered in the Cue List it will automatically calculate and display the end time. If the end time is blank, (i.e., : : :) the Cue List can not find the file in Open List. Make sure the sample is listed in Open List, and the two samples are spelled exactly the same.

c. **Problem** - The SMPTE source may not be set correctly.
Solution - If you are using an external SMPTE source, like a VTR make sure the internal SMPTE gen is closed. Also check the SMPTE source in Preferences. It should be set to AD516#1 or AD1012#1. If you are triggering from the internal SMPTE gen, make sure it is loaded and it running.

d. Problem- Too many errors in the incoming SMPTE time code. (Studio 16's internal generator will not cause this problem.)
Solution - It may be a bad cable. You can also open SMPTE monitor and watch for flashing squares in the upper left corner, they signify the time code errors. Error detection can be turned off from the Cue List's Prefs Option. See the Reference Section on Cue List for details.

e. Problem- The sample has been moved to another directory since the creation of the Cue List.
Solution - Move the sample back to the directory where it was when the Cue List was created. Or, change the directory for the sample in the Cue List by replacing the sample in the Cue List with the newly relocated sample from the Open List.

VII. Flashing Screens, or skipping, missing or repeating sound

Playback stutter or skipping occurs when your Amiga can't access the hard disk's data fast enough, or doesn't have enough CPU time to service the sounds that are currently being played or recorded. When Studio 16 detects this problem, it will flash your screen. The following are common causes and solutions for "skipping" audio. Although the causes are listed individually, "skipping" is often caused by multiple problems compounding one another rather than just one cause.

a. Problem - Your DMA mask setting on your hard disk may be incorrect. If your hard disk is abnormally slow or has an inability to playback 1 or 2 tracks, you may need to change the DMA mask for your hard disk.

1. **Solution** - If you have an A3000 or an A2000 with CBM controller, use 'HDToolbox' in your 'Tools' drawer. Once HDToolbox is running, select "Partition Drive" then "Advanced Options". Then select your partition from the "bar", and select "Change file system for partition". Make a note of the original MASK then change the MASK to: `0x7FFFFFFE`. To save the changes, select "OK", "OK", "Save changes to drive". You will need to re-boot your Amiga for the changes to take effect.
2. **Solution** - If you are using a GVP hard disk controller, use "FastPrep" in manual mode, instead of HDToolbox. See your GVP manual for more information. The correct DMA MASK for GVP controllers varies--call GVP. However, on newer controllers the above mask should work.

b. Problem - Your Amiga doesn't have enough CPU time to complete all the tasks required by Studio 16 to play the requested sounds.

1. **Solution** - Turn off CPU intensive Studio 16 modules, such as Meters, and SMPTE Monitor. Substitute Tiny Mixer instead of Mixer.
2. **Solution** - Upgrade your Amiga to a faster processor. If you have an Amiga 2000, install a 68030 accelerator card with fast RAM. For example, a stock A2000HD with the AD1012 doesn't have enough CPU power to play 4 tracks at 44K sampling rates. It can handle one or two tracks at 44K or four tracks at lower rates, such as 15K. (The AD516 will play 4 tracks without an accelerator.)
3. **Solution** - Using Transport, combine samples to be played back simultaneously into one sample. See Transport Reference Section on mixing for more details.
4. **Solution** - Upgrade your hard disk controller to a more efficient controller. The amount of bus resources required to transfer the same data to or from your hard disk can vary considerably between controller manufacturers. By switching to a more efficient hard disk controller, you leave more time for the CPU to work. Probably the most efficient hard disk controller currently available is Zeus 040 by Progressive Peripherals. Since the controller is built onto the 040 board, it writes data at a high speed and 32 bits at a time. This results in a significant performance increase.
5. **Solution** - Upgrade from an AD1012 to an AD516. SunRize's 16 bit card includes extra circuitry that makes it more efficient at transferring data than the AD1012.

c. **Problem** - The hard disk is too slow for the number of samples playing.

1. **Solution** - Reduce the number of simultaneously playing samples. If you can reduce the number of tracks playing simultaneously, you will reduce the data transfer rate and fix the problem.
2. **Solution** - Lower the Sampling Rate. If you lower your sampling rate, you can will reduce the amount of data that must be transferred and improve over all system performance.
3. **Solution** - Increase the Channel Buffer size in Preferences. By increasing the playback buffer sizes, you will eliminate skipping in many cases. However, the improvement will be less as you keep increasing the buffer sizes. That is, increasing the Channel Buffers from 256K to 512K will have much more of an improvement that increasing the buffers from 512K to 1024K. Keep in mind that one channel buffer is allocated for each playing sample, and each sample preloaded in Cue List. Make sure that you have enough system RAM to cover your requested buffers. For example, if you play four simultaneous tracks with the channel buffer set to 4096K, you will need 16,384K (or 16 MB) of free RAM just for buffers. For more about RAM requirements, see the Preferences Reference Section.
4. **Solution** - Buy a faster hard disk.

d. **Problem** - The hard disk is fragmented.

1. **Solution** - Fragmented hard disks are discussed in Chapter 4. The solution is to buy a program that will de-fragment your hard disk. Talk to your Amiga dealer for a recommendation.

e. **Problem** - The 'Use Extended Memory' option in the Preference Menu is set incorrectly.

1. **Solution** - Select this button if your hard disk controller and '030/'040 card are on the same PCB. (e.g. You have a stock A3000, or you have a GVP 030 card with built in hard disk controller.) Do NOT select this button if you have an '030 or '040, but your hard disk is on a stand alone Autoconfig card. (e.g. You have a GVP series II DMA hard disk controller and a PP&S 040 card with 32 bit RAM.) See the Preference Reference Section for details.

f. **Problem** - There is too large of a non-destructive edit in a sample.

1. **Solution** - Select Make Permanent from the Editor Menu.. When Studio 16 is playing a sample and encounters a non-destructive edit, it must seek over the edit. For large edits, this can take a long time and causes skipping.

VIII. Full Hard Disk

a. **Problem** - Running out of hard disk space is a problem for everyone eventually, some sooner than others.

1. **Solution** - Delete unnecessary data.
2. **Solution** - Make non-destructive edits permanent on samples that you have performed non-destructive cuts. Note that Make Permanent temporarily requires disk space equal to the file being made permanent. And, do NOT make samples with non-destructive paste-inserts permanent, unless you're prepared to tie up more disk space.
3. **Solution** - Buy another or a larger hard disk.
4. **Solution** - Reduce your sampling rate or resample your audio with a new rate in the Editor.
5. **Solution** - Record or Convert samples that don't require a high SNR, like explosions, to an 8 bit file (IFF-8SVX).

6. **Solution** - Archive your samples on a tape backup or an alternate backup device and then delete them from the hard disk.

IX. Gain won't adjust

a. **Problem** - Gain chip is malfunctioning.
   Solution - The Digital Pot in the input gain circuit may be bad. Suspect this if you can barely hear the input source or changing the gain level has no effect on the input volume. Call SunRize technical support if you suspect this problem.

X. Graph does not match sound

a. **Problem** - Graph file has been corrupted.
   Solution - Delete the sample's `.graph` file from the audio directory that contains the problem sample. You can delete files with WorkBench or Shell.

XI. Hard Disk Read/Write errors

Hard drives aren't fool proof, and it is inevitable that you will at some point encounter a disk error. This can manifest itself as a "Read" or "Write" error. Or, occasionally as a "Can't Validate Drive" error. Sometimes clicking "retry" or "cancel" on the DOS requester presenting the error will cause the error to disappear. However, if this happens you should start to worry because the error will probably reappear later, and can mean a failing hard drive.

Of course your best advice is to backup your hard drives. They can be backed up to a SCSI tape drive or a SCSI optical drive. However, due to the time and expense involved, many people don't bother. Just be warned that a hard disk "crash" is not just a possibility, it is very likely that it will happen to you eventually.

a. **Problem** - The hard disk gets intermittent read/write errors. - Some versions of the 1.3 FastFileSystem have bugs that cause Hard Disk Read/Write errors or crash when used with Studio 16. These problems appear to be more pronounced when using a SyQuest drive with a GVP series II controller.
   1. **Solution** - Upgrade your GVP ROM's to the latest version.
   2. **Solution** - Upgrade to WorkBench 2.0. It is much less buggy than 1.3
   3. **Solution** - If you must use 1.3, upgrade the FastFileSystem to a more recent version. We have included a recent version designed to work with 1.3 in the 'L' directory on the Studio 16 Master Disk #2. Copy it onto your hard disk with the SHELL.

```
Copy Studio16_2:L/FastFileSystem L:
```

   In order for the FastFileSystem to take effect, it must be copied to the Rigid Disk Block (RDB). This is done with the same software you use to partition your hard drive. NOTE: You must run your hard drive partitioning software (e.g. HDToolBox) after installing WorkBench 2.0 or a new FastFileSystem in order for the new file system to take effect. Contact your dealer or your hard drive controller manufacturer for more information.

b. **Problem** - The hard disk has crashed.
   1. **Solution**- Take your Amiga to your Amiga dealer. Their service department should be experienced in attempting to restore data from corrupt hard drives.
   2. **Solution** - Use a hard disk utility program to try and fix the errors on the hard disk. This may or may not work, but if you have any amount of time invested in the data on the bad partition, it's worth a try.

3. **Solution** - Use a program like "DiskSalv" to recover the files from the corrupt partition. "DiskSalv" is a public domain program that will copy files off a partition with errors onto a good partition. Once the operation is completed, you can re-format the bad partition and copy the files back over. However, DiskSalv requires a good partition or drive with enough free space to hold the copied files.
4. **Solution** - Re-format the partition that the error occurred on. This unfortunately erases all data on the partition being formatted. You can do this with the WorkBench "Initialize" option. Hopefully, you have a backup. If you do, restore the backup after formatting the partition.
5. **Solution** - If you don't have a backup, you can attempt to copy the data from the bad partition to another partition using WorkBench, DiskMaster/Opus, or the SHELL copy command BEFORE re-formatting the bad partition. This will probably recover some of the files, if not all. However, it does require a good partition or drive with enough free space to hold the copied files.

XII. Some Modules Don't Open When Selected

a. **Problem** - This is usually caused by a missing system library in the libs: directory. If a Studio 16 library is missing, you will get a warning (e.g. "Can't Open Studio.library"). However, if certain operating system libraries are missing (e.g. a math library) programs will simply not run.

Solution - Copy the libs directory from your master WorkBench floppy into your libs: directory on your hard disk. Insert you WorkBench Master disk into DF0: and from shell, type:

```
copy DF0:libs libs:
```

XIII. Samples loosing sync in Cue List or Transport

a. **Problem** - Samples have different sampling rates.

Solution - Multiple samples that playback simultaneously **must** have the same sampling rate. You can alter the sampling rate of a sample in the Editor. From the Effects Menu select the Resample option. Note that if the sampling rates are drastically different, you'll hear one sample play at the wrong speed. However, if the sampling rates are only slightly different, the samples will sound right, but you'll loose sync as the samples play.

Chapter 8

Third Party Sources

This chapter contains a list of third party products that you may find useful when using Studio 16. We have only tested a few of these products. **We are not recommending any of these suppliers, they are provided for your convenience only.**

SMPTE VITC to LTC Converters	Horita P.O. Box 3993 Mission Viejo, CA 92690 USA (714) 489-0240
SMPTE LTC Generators **MIDI to LTC Converters**	MidiMan 30 N. Raymond Pasadena, CA 91103 USA (818) 449-8838
Studio 16 Compatible MIDI Sequencers	Bars&Pipes Professional The Blue Ribbon SoundWorks, Ltd. 1605 Chantilly Dr., Suite 200 Atlanta, GA 30324 USA (404) 315-0212
Video Effects Generator, Switcher, Paint Box, and 3D Modeler	Video Toaster NewTek Inc. 215 E. 8th St Topeka, KS 66603 (800) 765-3406

Production Music	Killer Tracks 6534 Sunset Blvd. Hollywood, CA 90028 (213) 957-4455 (800) 877-0078
	Philadelphia Music Works P.O. Box 947 Bryn Mawr, PA 19010 (215) 825-5656 (800) 368-0033
	Valentino 151 W. 46th St. New York, NY 10036 (212) 869-5210 (800) 223-6278
Sound Effects Libraries on CD	Sound Ideas 105 West Beaver Creek Rd., Suite #4 Richmond Hill, Ontario L4B 1C6 CANADA (416) 886-5000
	Valentino 151 W. 46th St. New York, NY 10036 (212) 869-5210 (800) 223-6278
Removable Media Drive	SyQuest Technology 47071 Bayside Parkway Fremont, CA 94538 800-245-2278
Magneto Optical Drives	Ricoh 3001 Orchard Parkway San Jose, 95134 800-955-3453
	Panasonic Comm. and Systems 2 Panasonic Way Secaucus, NJ 07094 800-742-8086
VHS Deck modification that allows simultaneous access to Hi-Fi and Linear audio tracks. Decks include AG-1960.	Carlson-Strand Company P.O. Box 3761 San Clemente, CA 92674 (714) 492-8978

Module Reference

The Module References detail each Studio 16 module. Included is the name, class, and description of every module and command and a detailed explanation of all gadgets and menus. Some modules also include step by step instructions for common procedures.

Name:	The name of a module as it appears in the Application or Project Menu.
Keyboard Shortcut:	A keyboard equivalent is available for launching application modules by holding down the control (Ctrl) key and typing the indicated letter.
Class:	
Application	Studio 16 program that performs a specific function for the user, e.g. Recorder.
Device Driver	Studio 16 module that communicates between the software and the digitizing/playback hardware, e.g. AD516Handler
Utility	Studio 16 program that provides services to Application modules, e.g. DiskIO.
Shell Command	Studio 16 program that can be executed from the Shell to perform a specific function, e.g. StudioPlay.
Description:	Describes the function of the Module or Command.
Layout:	Describes the appearance of the Module, generally including a screen shot.
Format:	Lists the required format to execute a Shell command.
Procedures:	Step by step instructions for the most common applications of a module.
Gadget Reference:	Detailed reference to each button and gadget for a module.
Menu Reference:	Detailed reference for the menu options of a module.

Name: **AD516 Handler**

Class: Device Driver

Description: The AD516Handler is responsible for handling data transfer to and from the AD516 card. There should be one AD516Handler resident for each card you have installed in your Amiga. In general, you don't need to access the AD516Handler. However, if you have installed multiple AD516 cards, you may need to use ModList to load another AD516Handler.

Procedures:

Check Hardware Revision Numbers & Statistics

1. Select Instance from the Applications Menu (^ I).
2. Select **Show Utilities** from the Instance Menu. The Instance List will update showing all the drivers and utility modules.
3. Double click the **AD516Handler** entry in the Instance List. An AD516Handler display window will open with various version numbers and the amount of static RAM installed in your AD516 card.

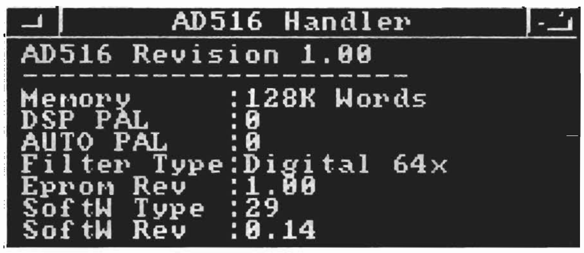


Figure 1 - Typical AD516 Handler

Name: **AD1012 Handler**

Class: Device Driver

Description: The AD1012Handler is responsible for handling data transfer to and from the AD1012 card. There should be one AD1012Handler resident for each card you have installed in your Amiga. Installing more than one AD1012 is not recommended.

Procedures:

Check Hardware Revision Numbers & Statistics

1. Select Instance from the Applications Menu (^ I).
2. Select **Show Utilities** from the Instance Menu. The Instance List will update showing all the drivers and utility modules.
3. Double click the **AD1012Handler** entry in the Instance List. An AD1012Handler display window will open with various version numbers and the amount of static RAM installed in your AD1012 card.

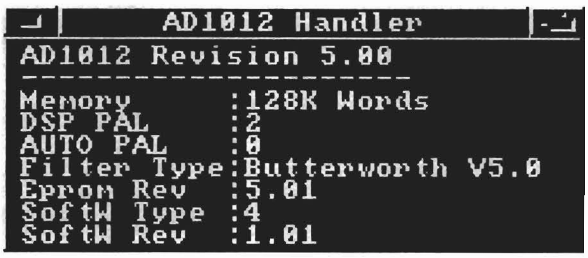


Figure 1 - Typical AD1012 Handler

Name: **Cue List**

Keyboard: ^C

Class: Application Module

Description: The Cue List module allows you trigger samples from SMPTE time code. SMPTE time code is a time stamping format used in video production. Each video frame has a specific time code associated with it. You can instruct Studio 16 to trigger a sample on any video frame. This gives Studio 16 powerful audio for video editing capabilities.

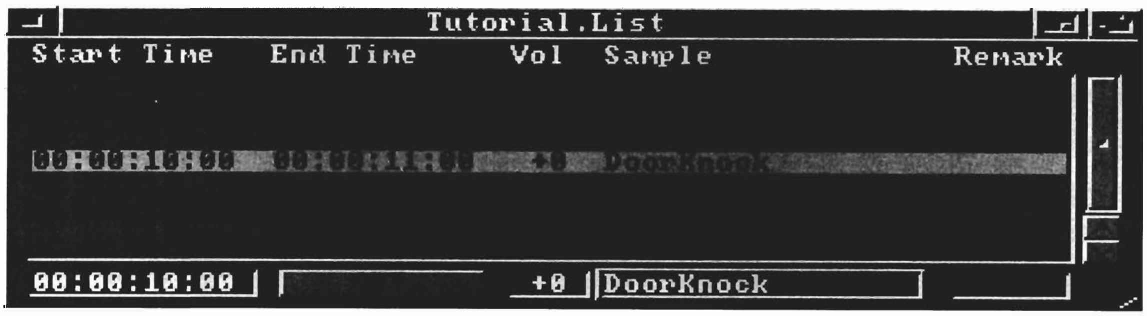


Figure 1. Cue List

Layout: The Cue List window is comprised of three main elements. The List itself takes up the majority of the window space. It displays each SMPTE event that Studio 16 will trigger. Just below the Cue List is the Edit Line. The Edit Line is used to alter existing entries in the Cue List. At the bottom of the window is an ON/OFF button. Figure 1.

Procedures:

Entering Samples

The are two methods for entering samples into a Cue List. The simplest is to click and drag a sample name from the Open List or from the Editor Regions List into the Cue List. This will automatically cause the SMPTE time code to update and you can be sure the sample is available for playback.

The second method is to select Add from the Entry Menu. You can then type in a sample's name from the file requester and hit return to add it. Note that the name must be in the Open List and spelled exactly the same. With both methods, you can enter a volume setting and a remark.

> NOTE: If a sample has a SMPTE start time associated with it, you must drag that sample in from Open List. Cue List will not be able to update the SMPTE Start time if you type in the sample name in the Edit Line, (method 2).

Edit an Entry

Highlight an existing entry by selecting an entry in the Cue List. The entry highlighted will appear in the Edit Line at the bottom for editing. Type in any changes to the Start

Time, Volume or Remark, you can also change the sample all together by entering a new Sample name. Make sure the new sample name matches a sample in the Open List. Always hit return after making a change in the edit line to update the Cue List.

To replace one sample with another, highlight the sample to be replaced so it shows in the edit line. Then, drag a new sample from the Open List or Editor Regions List and drop it in the Edit Line. This will replace the original entry.

Delete an Entry

Select the appropriate entry in the list and select **Delete** from the Entry Menu, or type *A*-D. This doesn't delete the file, it only removes the entry from the Cue List file.

Creating a Cue List with a Text Editor

You can also create and edit Cue List files using your favorite text editor. Simply load a Cue List file (produced by the **Save** button in the Cue List module) into any word processor or text editor and type in the changes. The correct format is as follows:

TQL↵
Start Time ,Volume ,Sample Name ,Remark↵

Example:
TQL↵
00:02:01:00 ,+1 ,FootSteps ,Mike Leaves↵
00:02:30:00 ,0 ,pan 200, DoorSlam↵
00:03:04:00 ,-3 ,pan 150, CarStartAndGo ,Porshe 911↵

The first line of the Cue List must be 'TQL' immediately followed by a hard return (↵) for Studio 16 to recognize the file. The first character of text must immediately follow the comma or start in the first column. Any number of spaces may follow the text and a remark is optional. Pan will appear in the Remark field, (see Pan reference later in this section). The end of each line should be followed by a hard return.

Adjusting Start and End Times

You can change the time code, by deleting the current time and typing in a new one. Always hit return to update the Cue List.

A Shortcut for Changing the Start Time - delete the time in the start field and type +3. This will add 3 frames to the original setting. To make the sample trigger 15 min., 8 sec., and 12 frames earlier. Delete the current setting and type -150812 in the start time field, and hit return. You can omit the leading zeroes and colons " : ".

The end time is automatically calculated from the length of the sample and start time. The end time of a cue event may be altered by changing the start time for the entry or the length of the entry's sample in the Editor. If a blank End Time is displayed (: : :), Cue List has not found the sample in Open List. Check Open List to make sure the sample is loaded, and it is spelled correctly in the Cue List.

Note: The recommended minimum Start Time is 1 second. This allows Studio 16 time to preload the sample before it has to play it.

To trigger part of a sample instead of the entire sample, use the Editor to create a region and drag the region's name from the Editor Regions List into the Cue List.

Loading and Saving Cue Lists

The Cue List Menu options, **Load** ,**Save**, and **Save As** are used to store and retrieve Cue List files. To load an existing Cue List select the **Load** button. A File Requester will open allowing you to choose a Cue List file to load. To store a Cue List for later use, select the **Save As** button. A File Requester will open asking you for the name of the Cue List file. Type in the name of the file you wish to save and then select the **OK** button.

Note: A Cue List file does not contain samples, it is a text file that contains descriptions of cue events to be triggered.

Triggering the Cue List

The Cue List must be **ON** for samples to trigger. Also, the SMPTE generator (figure 2) must be running, or an external source has to be plugged into the AD516 or AD1012 and running.

To play an individual sample, select Play from the Entry Menu.

Triggering from the middle of a sample is useful when you are working with long samples and it is more convenient for you to start time code in the middle of a long sample. Generally samples are triggered on the specified time code. But if time code starts or jumps to the middle of a sample, Studio 16 will attempt to trigger the remaining portion of the sample. It may take Studio 16 several seconds to find the correct position to start playback, especially if the computer is busy, or if the sample is long and you are triggering near the end.

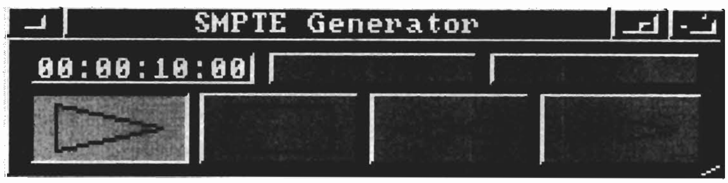


Figure 2. SMPTE Generator

Selecting SMPTE Frame Rate and Source

Use Preferences from the Project Menu to choose your SMPTE frame rate and default source. This generally only needs to be set once.

Studio 16 supports SMPTE frame rates of 24, 25, and 30 frames per second. Make your selection and select SaveSetup from the Project Menu. (More details are available in the Preference Reference Section.)

In most cases, the default SMPTE source will be correct. The default should be either the AD516 or the AD1012. So whenever time code in coming into the card, SMPTE monitor will display the frame count if your external source, provided it's turned on.

When you work with the Studio 16's SMPTE Generator, Preferences will override the default setting with SMPTE gen.

Gadgets:

List The current list of samples set to be triggered. Displayed is the start and end time, volume, sample name, and an optional remark for each entry. Entries are generally dragged in from the Open List or the Region List in the Editor.

Start Time The time the sample will begin playing. Units are in SMPTE format - HH:MM:SS:FF (hours:minutes:seconds:frames). Suggested minimum Start Time for each sample is 01:00. This allows Studio 16 time to preload a sample before having to play it. The start time can be altered by changing it in the edit field or deleting the time and typing in an incremental change like +3 to add 3 frames the existing time.

End Time The calculated time the sample will stop playing. It is not adjustable. Use the Editor to create a smaller region for samples that are too long.

Vol The selected Volume (in decibels) at which the sample will play. Entries can range between +06dBs to -99dBs.

+06 dBs = 200% Volume
0 dBs = 100% Volume
-06 dBs = 50% Volume
-12dBs = 25% Volume

Sample The name of the sample. It must match an Open List entry exactly.

Remark Optional note attached to a Cue entry. Will also accept commands for setting the pan.

Pan **AD516 ONLY** You can set the pan for a sample by entering a special command in the Remark field. Pan allows you to vary the level of audio to playing on each channel. The AD516 has two channels, and you can set a sample to play on either the left, right, or both output jacks. When pan is set to 100, the sample will play equally on both the left and right output. If the pan is set to 0, the sound will only play on the left output. If pan is set to 200, the sample will play only on the right. You can also vary the levels on each output. For instance if the pan is set to 103, the sound will be 3dBs louder on the right than on the left.

The following are examples of the format for setting a pan in the Remark field. Make sure the 'p' in pan is the first entry in the remark field.

pan 0	full left
pan 100	center (left and right equal)
pan 155	55dBs higher on the right
pan 200, door closes	full right

The usual remark like "door closes" can be entered after the pan setting. In the above sample the door closing sample would only be heard on the right channel, which is appropriate if the door is to the right of the viewer.

Samples recorded in stereo will automatically default to the correct pan setting. You can override this by entering a new setting in the Remark field or by resetting the default pan value in the editor.

Edit Line The update boxes at the bottom of the window show the name, start time, vol and remark for the highlighted sample. An entry in the Cue List can be edited by highlighting it and typing into the Edit Line fields. Always hit the Enter key to update the Cue List.

Menu: **Cue List**

New Select this option to clear the current entries.

Load Allows you to load a Cue List File. It brings up an Open File Requester at the default directory Studio16:CueLists. A Cue List file is a text file that contains sample names, vol., etc. In addition to loading a Cue List file, the actual samples must be in the Open List before they can be triggered.

The default directory can be changed, by selecting a new directory in the requester and then selecting Save Setup from the Project Menu.

Merge Allows you to merge two Cue List files. The new file will sort itself by start time.

Save Save the changes made to the Cue List with the current file name.

Save As Select this option to save the current Cue List with a new name. It brings up a Save File Requester at the default directory of Studio16:CueLists. The file saved consists only of sample names, start times, volume settings and optional comments.

Offset Allows you to enter a global offset for all entries in a Cue List. The offset can be typed is as a complete time code, 00:01:23:14, or without the leading zeroes and colons (12314), or as an incremental change +1202.

Prefs

Reaction (frames)
Sets the reaction time for TC Add and TC Update. If the start times you add with TC Add (*A*-T), are continually 2 frames late, change the reaction setting to -2 frames.

SMPTE Errors
When SMPTE Monitor is running, it is set to detect three types of errors. When these errors are detected, you will see flashing squares in the upper left corner of the SMPTE Monitor. Every time a square flashes, it means that a type of SMPTE "error" was detected. The detection of these errors may cause a sound to stop playing. Turn off the error detection by deactivating the error.

Each of the three squares in the SMPTE monitor represent a different error. The three squares from Left to Right are:

SMPTE Time Out [white] - No time code was detected for about 1/2 a second. Pausing your video deck will cause this.

SMPTE Time Code Error [black] - This error is generated if an illegal time code is read by the SMPTE reader. For example, a time code with 69 seconds would generate this error. This error typically results from distortion, usually a bad cable, a bad connection, or tape drop out.

SMPTE Time Code Jump [yellow] - This error means that a discontinuity occurred in the time code. For example 00:01:00:01, 00:01:00:02, 00:03:00:12, 00:03:00:13 would cause this error. Jumps typically occur when you fast forward or rewind the time code source.

Figure 3. Cue List Prefs

Preload
This is the number of entries that are preloaded into RAM before playback. The default for the AD1012 is 5. It should be adequate unless you are triggering a lot of short samples (samples under 2 seconds in length). If you are using the AD516 increase the preload number to 9. Note that for every sample preloaded you must have an available channel buffer. Example, if the channel buffer in Preferences is set to 256K, and you want to preload 8 samples, you must have at least 2 Megs free.

Menu: **Entry**

Add Enters a blank entry at 00:00:00:00. The blank entry can then be edited from the edit line. Or you can replace the blank entry by highlighting it and then dropping a new entry from the Open List or the Editor Regions List in the Edit Line. This will replace the black entry in the list.

Duplicate This button duplicates the highlighted entry in the Cue List.

Delete Click this button to delete the highlighted Cue List entry.

TC Add Adds an blank entry at the current time code. It allows you to select hot spots in a video by hitting (*A*-T) every time you see a frame you want to trigger audio on. You can later drop samples in on these time codes. Set a reaction time for TC Add with the Prefs Option in the Cue List Menu.

TC Update Changes the start time of the highlighted entry to the current time code when selected. You can set a reaction time in Cue List Prefs.

Play This button plays a single highlighted Cue Entry. To trigger the Cue List in sequence, activate your SMPTE source or the internal SMPTE Generator.

Stop Stops all samples playing back. To stop the playback of a Cue List that has been triggered, turn the Cue List OFF. Or, stop the SMPTE time code generation.

Name: **DiskIO**

Class: Utility Module

Description: Responsible for handling data transfer to and from your hard disk, DiskIO is not a module that requires user access.

Name: **Editor**

Keyboard: ^E or *A*-E*

Class: Application Module

Description: The **Editor** allows you to manipulate the data of sound files. It is accessed from **Applications Menu** or from the **Open List**. From the Applications Menu, load an empty editor by selecting Editor. To open an editor for a specific sample, highlight a sample in Open List and select Edit Sample. Note that multiple select works to edit samples simultaneously.

Editing is accomplished in Studio 16 much as it would be with a traditional 8 bit sample editor like Perfect Sound or Audition 4. An editor displays a graph of the sound being edited and allows you cut, copy, and paste sounds. You can move parts of a sound around, delete unwanted sections, fade in or out sections of sound, or even reverse sound. You can also use the "freehand" mode to draw with the mouse to remove pops. In addition, the editor also allows you to change a sample's default playback rate, volume, and filter setting and contains more advanced features like echo, FFT, and resample.

Figure 1. Editor Window

Layout:

Status Display

Along the top of the **Editor** is the status display. It includes numerical positions of the marked range and the current graph display. The numbers on the far left and right are the first and last positions for the entire display. The numbers toward the center of the window are the fist and last positions of the marked range. The length of the range is displayed in parentheses.

The status display is generally shown in SMPTE time code. It can be displayed in numbers of samples by activating Units in Samples in the Editor's Option Menu.

* ^E Opens an empty Editor.
A-E Opens an Editor with a sample loaded provided the sample is highlighted in the Open List and the Open List window is active.

The status display is generally shown in SMPTE time code. It can be displayed in numbers of samples by activating Units in Samples in the Editor's Option Menu.

You also have the option of entering an offset for the status display. The offset is added from the Editor's Option Menu - Set Display Offset. An offset is often used in conjunction with the Cue List. For example, when editing a sound that is also entered in the Cue List, you can set the offset equal to the Cue List start time code. So when you drag ranges, the SMPTE display of the ranges in the editor window will match the sound's position on tape.

Graph Most of the **Editor** window is occupied by a sound's graph. The vertical axis of the graph represents the amplitude of sound, and the horizontal axis represents time.

Because Studio 16 samples are hard disk based (they are not in RAM), the editor has a unique difficulty. How can a graphic representation of the digitized sound be displayed fast enough to be usable? Studio 16 solves this problem by building a graph in RAM from hard disk, and using this "graph buffer" to display the "on screen" graph.

When you first edit a sound, a window will appear and a graph will begin to form from the left to the right. Your hard disk light will also be quite active. Studio 16 is building its "graph buffer", a process that only has to be done once - when you first load the sample in the editor. For long samples, this can take a while. For example, a 3-4 minute sound (~14 MB) will take about 40 seconds to build the graph. This process can be sped up slightly by increasing the "Copy Buffer" in Preferences. Once the graph is built, the editor window will be redrawn with the graph displayed. For quicker graph updates, select the Fast Graph option in the Editor's Options Menu.

Editing is accomplished by dragging your mouse over the graph to mark a range, then selecting one of the editor menu options.

Buttons The navigation buttons are located along the bottom of the graph. See the Gadget Reference for a detailed explanation of each button.

Common Procedures:

Load a Sample and an Editor

1. Select Open List from the Applications Menu. (^O)
2. Select a sample to edit, and then select Edit Sample from the Open List Menu. (*A*-E)

Removing Leading Dead Space

1. Examine the beginning of the graph to note where the dead space, or silence appears.

2. Using the graph as a guide, use your mouse to drag a range highlighting the silence. This will be easiest if you click on the right side of the intended range and drag to the left. By dragging past the left edge of the graph, you will be sure your range includes the first sample.

3. Type *A-L* to play the marked range to listen to the area you have just highlighted (or select Play Range from the Editor Menu). You should not hear anything, except perhaps background noise.

4. On the basis of listening to the range, you may want to fine tune your marked range. You can use the mouse to grab the end of the marked range and adjust it. To get a closer view of your sample, click the Show Range button, and then click the Zoom Out button to zoom out a little.

5. Once the range is finalized, select Non-Destructive Cut from the Edit Menu. You have now performed a "non destructive cut." The data you have deleted is still on your hard disk, but will not be played when you access the sample. You can restore it by selecting **Undo Last**.

6. If you want to free the hard disk space currently being occupied by the silence, you need to make the non-destructive edit permanent. Select Make Permanent from the Edit Menu. This operation can take a while, depending on the length of your sample. Once it is complete, the graph will be regenerated.

Fading a Sound In

1. Examine the beginning of the graph to note where the fade should occur.

2. Using the graph as a guide, use your mouse to drag a range highlighting the section to fade. This will be easiest if you start on the right side of the intended range and drag to the left. By dragging past the left edge of the graph, you will be sure to start your fade on first sample.

3. Type *A-L* to listen to the area you've just highlighted.

4. On the basis of playing the range, you may want to fine tune your marked range. You can use the mouse to grab the end of the marked range and adjust it. Zoom In for a closer look of the waveform by clicking the Zm In button.

5. Once the range is finalized, select Scale from the Effects Menu.

> NOTE: Fades are permanent. Once done, they can not be undone. You may want to make a copy of the sound before doing a fade as a reserve. To make a copy of a sample, click the Range All button and then use the steps in the next section for copying a range to a new sample.

6. A Select Scale Requester will appear. To fade in, change **Start** to 0%, and **End** to 100%. Figure 4. To fade out you would set Start to 100% and End to 0%.

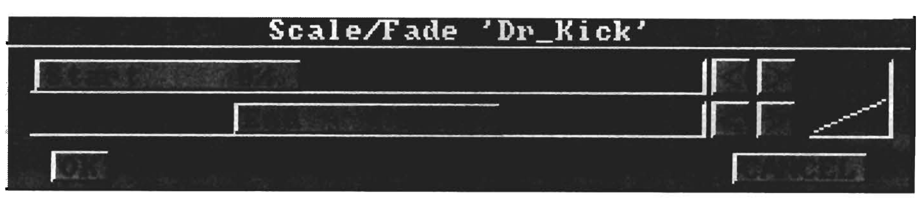


Figure 4. Scale Selector

7. Click **OK**.

8. The fade will now be performed. It may take a while, depending on the length of the marked range.

Creating a new sample out of a marked range

1. Highlight a range on the main graph. A new sample will be created out of this range. To duplicate an entire sample, click the Range All button.

2. Select Destructive Copy from the Edit Menu. A destructive copy will cause a new file to be created from the marked range called *CopyBuffer*.

3. Bring the Open List module to the front by typing ^O, or by selecting it from the Applications Menu.

4. Select the sample *CopyBuffer* in Open List.

5. Select **Rename** from the Open List Menu and type in a new name for the sample and hit the **Enter** key.

Pasting a Range from One Sample into Another

1. Before you begin this tutorial, close all modules except **Open List**.

2. Pick two samples in the Open List that have the same sampling rate. They will be referred to as *SourceSample* and *DestinationSample*.

3. Highlight one of the samples in the Open List and type *A*-E, or select the Edit Option from the Open List Menu.

4. Bring Open List to the front (^O). Highlight the second sample and select Edit again to open a second Editor.

5. You now have 2 editors open. Resize and move the Editor windows so that you can view both of them on your screen at once. Figure 5.

6. Mark a range on the graph of the *SourceSample*. This will be pasted into *DestinationSample*.

7. From the *SourceSample* editor, select Destructive **Copy**. At this point, a new sample will be created called *CopyBuffer*.

8. Mark an insertion point for the range in the *DestinationSample*.

9. In the *DestinationSample* Editor, select Destructive Paste - Insert at Start. *CopyBuffer* will be inserted now. The process can take a while if the files are large.

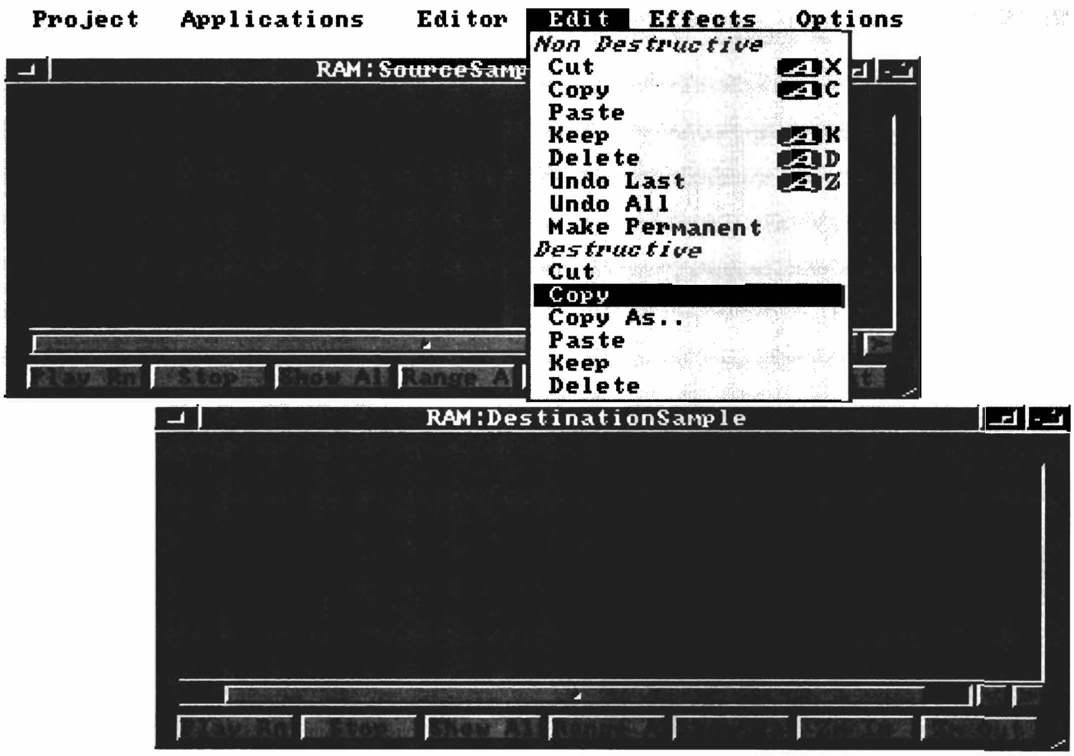


Figure 5. Moving a Range from *SourceSample* to *DestinationSample*.

Buttons:

Play Range This will play the marked range in the editor window.

Stop Any playing sound is stopped.

Show All Adjusts the zoom ratio so that the entire sample is displayed on the graph.

Range All Will create a range over the entire sample. Will also Show All in the process.

Show Range This expands the marked range to fill the graph - changing the zoom ratio and the scroll as needed.

Zm In To "Zoom in" on your sound's graph to see more detail, click this button. Successive clicks will zoom you in closer until you are zoomed all the way in. When zoomed all the way in, you will be viewing the sample on a scale where each horizontal screen pixel represents one sample.

Once you've zoomed in on a sample, you can use the scroll bar along the bottom of the graph to move around in the sample. Usually the graph is updated from the "graph buffer", so scrolling is a very quick process. However, if you zoom all the way in, the editor may access the hard disk. When this happens, the updates will become slower, and you will see the hard disk light flash. To speed things up, zoom out a little.

You can also zoom in on the sound by marking a range (use your mouse to drag a range on the graph) and then clicking the **Show Range** button. This will expand the marked range to fill the graph.

Zm Out Reverses the zoom ratio, so you can see more of a sample. Click Show All to zoom all the way out of a sample quickly.

Menu: **Editor**

New Clears all samples from the Editor Window. The samples are not deleted, they are just removed from the Editor.

Open Opens a file requester allowing the addition of a sample to the display. Up to 8 samples can be in the editor at once allowing synchronous edits.

Play All Plays the entire sample in the editor. You can also play sounds from Open List, Transport and Cue List.

Play Range This will play the highlighted range on the graph.

Play Display Plays the portion of the sample that is currently displayed in the graph.

Play to Start Plays a small portion of a sample preceding the marked range.

Stop Playing All playing sound is stopped.

Menu: **Edit**

Non Destructive

A "non- destructive" edit will not alter you original sound. It just remembers which changes have been indicated. Since non-destructive edits don't alter your hard disk data, they can be "undone" at any time to restore your original data. Another benefit of "non-destructive" edits is speed. Since hard disk access is not required, non-destructive edits are much faster to execute than destructive edits.

Non-destructive edits only work on one sample at a time. That is, you can't do a non-destructive copy, then paste the copied range into a different sample. The non-destructive paste must be into the same sample where the non-destructive copy was performed.

Cut This is the same as selecting a non destructive **Copy** and **Delete**. If you want to delete a range while maintaining the sound after the cut in sync, use **zero**.

Copy This remembers the marked range for use in future pastes. No new sample is created.

Paste Non-Destructive **Paste** has a submenu that determines where to paste:

Insert at Start
Insert at End
Replace Range

Selecting **Insert** at Start or End will insert the copied range at the beginning or end of the marked range. Selecting **Replace** Range will replace the marked range with the last copied/cut range.

The non-destructive **Copy - Paste - Insert** action allows you to duplicate a section of sound repeatedly. You can insert multiple copies of a range into a sample without using more hard disk space (unless you select Make Permanent or do a destructive edit).

NOTE: With Studio 16 version 2.0 there is a limit of 128 non-destructive edit clips.

Keep This edit is the inverse of **delete**. It deletes all of the wave except what is highlighted in the marked range.

Delete This deletes a marked range (it can be reversed by **Undo**). To delete a range while maintaining sync, use **zero**. (NOTE: Zero is destructive.)

Undo Last This will undo the last non-destructive edit. It does not have any effect on other types of edits or effects. Clicking **Undo** a second time will reverse the Undo.

Undo All This option will undo all non-destructive edits. It has no effect on destructive edits or effects.

Make Permanent

Allows you to do non-destructive editing on a sample and then transform the results into a permanent sample with a single "destructive edit". This option will update the current sample's data taking non-destructive edits into account. Performing a destructive edit will also make all non-destructive edits permanent.

There are two reasons to make non-destructive edits permanent:

1. You have performed non-destructive cuts on a sample so that if the edits were made permanent only a fraction of the disk space would be required by the sample.

   Example - You record a 20 MB sound file, then edit the sample in the non destructive mode, doing cuts, copies and pastes. The resulting sound would fit in a much smaller 5MB space. However, since your edits were non-destructive, the original 20MB file is still on your hard disk. If you're confident you won't need the original unedited sample, by making the edits permanent you can create a new sample that contains just the 5MB of edited data, thus, freeing 15MB of hard disk space.

2. You are experiencing long delays between selecting **Play All** or **Play Range** and the sound actually playing. This happens when the computer must seek long distances from the beginning of the sound's data file to find the point where playback begins. AmigaDOS seeks can be quite slow. As a result, you may want to make the edits permanent to eliminate the delay.

NOTE: Make Permanent will create a new file on your hard disk, then delete the old file. As a result, it will not work unless there is enough free space on your hard disk to temporarily hold both the original file and the new "compressed" file.

> NOTE: Make Permanent can make a file larger if you have performed many **Paste - Inserts**.

DESTRUCTIVE

An edit is "destructive" if it alters your sample's data on the hard disk. Destructive edits are generally used to transfer clips from one sample to another. (Non-destructive edits only work within a single sample.) For example, if you wanted to cut part of sample and paste it into an another sample, you would have to use Destructive Copy and Paste.

Destructive edits are also used when you want to create a new sample from an existing sample.

As well as the destructive **Cut**, **Copy**, and **Paste**, effects are also destructive. These include **Resample**, **Scale**, **Invert**, and others.

Cut A destructive cut will remove a marked range and place it in a sample called CopyBuffer. *CopyBuffer* is a normal sample in all respects, except that by convention it is used in Destructive **Paste**. CopyBuffer is created in the active directory.

Copy Destructive copy will copy a marked range into a new sample called *CopyBuffer.*

Copy As It is often useful to create a new sample out of a marked range. Copy As will bring up a file requester and prompt you for a name for the new sample.

> NOTE: Copy As will create the new file with the default file format set in Preferences. This allows you to easily create samples in different file formats from ranges of original samples.

Paste Destructive **Paste**, like its non destructive counterpart, has a sub menu that allows you to select where to paste:

Insert at Start
Insert at End
Replace Range

Selecting **Insert** at Start or End will insert any sample named *CopyBuffer* at the beginning or end of the marked range. Selecting **Replace** will replace the marked range with the contents of *CopyBuffer.*

To transfer a clip of sound between samples, do a Destructive **Copy** in the source sample followed by a Destructive **Paste** in the destination sample.

To append one sample to another, select Insert @ End. But make sure the destination sample's range includes the last sample of the sound.

Delete Delete will remove a range from the sample. A delete is similar to a cut, but it does not create a CopyBuffer, and it is quicker to perform. It may be preferable to delete a range, if you are working with long samples and you are certain you do not require the marked range. There is no undo for delete.

Keep Keep is the reverse of delete. It will remove all of the sample except for the marked range. It is safer to do a non-destructive keep and then make the edit permanent.

Menu: **Effects**

Echo This option brings up an echo range requester that allows you to set the Delay, Feedback and DLevel of an echo. These settings are similar to the Realtime Delay Parameters. Try will preview the effect; however, it requires an accelerated Amiga.

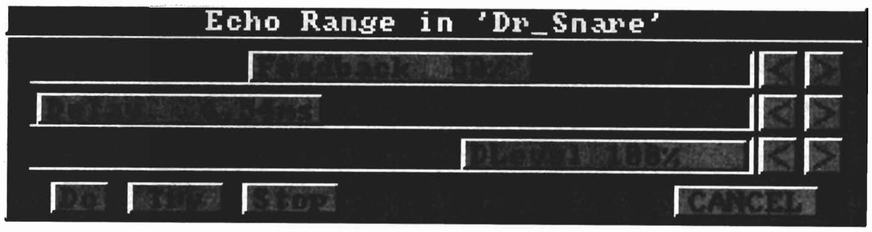


Figure 6. Echo Selector.

Delay Adjusts the length of time between repeats.

Feedback Controls the number of echoes you hear. This is accomplished by adding a percentage of the delayed signal back into the original.

D Level Controls the volume of the echoes, at 100%, the first echo and the original signal are the same volume. At 0% you will not hear an echo.

FFT Fast Fourier Transform breaks down a waveform into its constituent sinusoidal parts. The "Fourier Transform" is a mathematical transform that translates time domain information (like sampled sound) into the frequency domain. The "FFT", or "Fast Fourier Transform" is a computationally efficient method of computing the Fourier Transform. Actually, Studio 16 implements the "DFT" or "Discrete Fourier Transform". However, many people use FFT and DFT interchangeably.

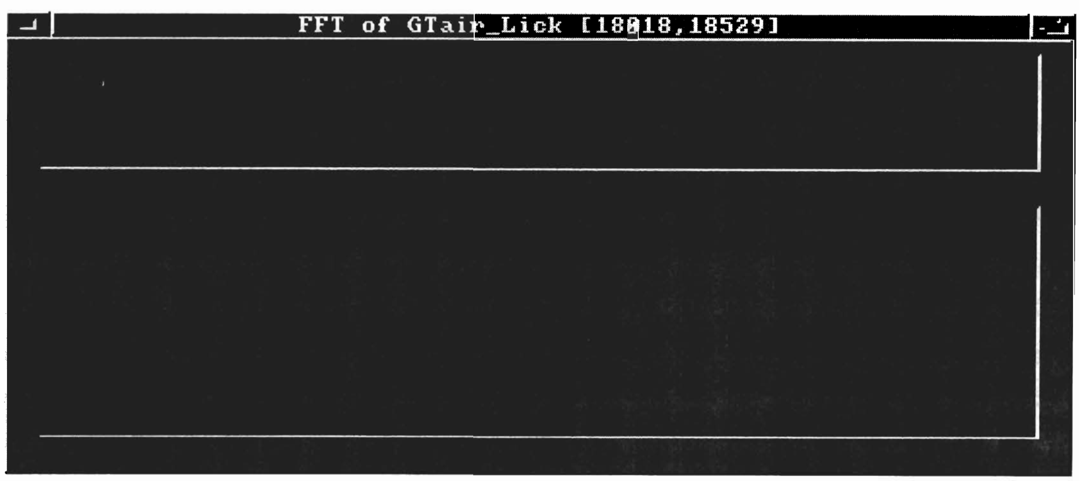


Figure 7. FFT Display

In any case, the FFT will produce a plot that shows you which frequency components are present in the samples analyzed. An FFT algorithm always processes a power of 2 samples. In Studio 16's case, it always processes 512 samples. It then produces 256 real "frequency samples". Each of these points is plotted by Studio 16 in a bar graph. Since there always 256 points, or "bins", these bins contain a range of frequencies. The center frequency in this range is displayed in the upper right hand corner as you move the cursor.

When selecting a range for an FFT keep in mind that only the first 512 samples will be used in the calculation. To mark a range that is exactly 512 samples in length, activate Units in Samples from the Option Menu in Editor. Then, watch the range's status display (the center most numbers along the top of the graph) as you adjust you range to 512 samples.

Invert This option inverts the ranged waveform along the y-axis. The sample will not sound different.

Normalize A variation of scale that first measures the peak value of the sample, and then scales the entire range by the percentage required to bring the peak to a maximum setting. The maximum level for 16 bit samples is 32,767. Refer to Chapter 1 for more information on maximum levels.

Normalize can also be used to find the maximum sample in a range since it is determined before scaling and you can always cancel the normalization before it alters your sample.

Reverse This option will reverse a sound's data so that it plays backwards.

Scale This option allows you to scale (attenuate or amplify) a sample. It also allows you to fade in or fade out a section of a sample by specifying a starting and ending percentage. When they are equal, a straight scale is performed. When the percentages differ a fade will be performed. By combining Editor Fades with the Cue List, cross fades can be performed.

When using scale to increase the level of a range at an even rate, consider using Normalize.

Studio 16 Scales are linear. To produce a logarithmic scale, scale the same range twice at the same setting.

Zero Mark a range and select zero to silence the existing audio. This will allow you to remove sound from a track while maintaining sync throughout the entire track.

Resample Allows you to adjust the sampling rate of a sample. The parameters will affect the quality of the sample and the amount of RAM required. Also, resampling is a complex operations, so it may take a while, especially if you are working with long samples.

Resample can also be used to "pitch shift". Resample a sound to a new rate, and then playing it back at its original rate.

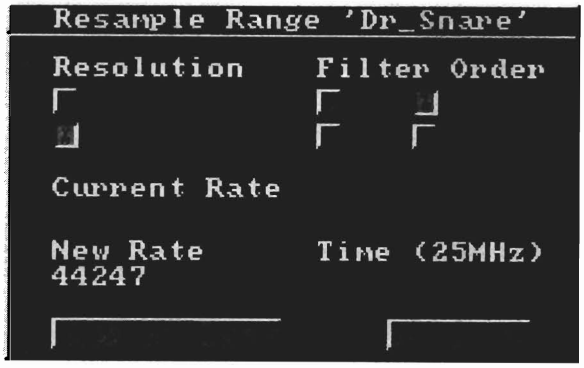


Figure 8. Resample Selector.

Resolution Because resampling is very calculation intensive, Studio 16 pre-computes a sinc table for speed. "Resolution" refers to the size of the sinc table. The normal table is provided for machines with small amounts of RAM. High Resolution requires more RAM, but may give slightly better results because it uses a larger table.

Filter Order Resample is controlled by a digital filter. Digital filters allow you to set an order of the filter or the degree of accuracy. The larger the order of the filter, the more accurate the resample will be. However, high filter orders take longer to process. The Filter Orders that are available for resampling are 5, 11, 21, 101.

Current Rate The default sampling/playback rate of a sample before resampling.

New Rate A field for entering the new sampling rate. (Open the Recorder to see the available rates your AD516/AD1012 can playback)

Time The estimated time required to process the resample. Note the field will update as you select different filter orders. This estimated time is for an Amiga 3000 with a 25MHz processor. Stock Amiga 2000s will take much longer.

Gen Silence This option will create silence in the CopyBuffer. A requester will appear allowing you to select the length of the silence and the sampling rate. Once the silence is in the CopyBuffer, you can select destructive edit and paste the silence into your working sample.

Sampling Rate The default sampling rate is from your working sample. If you intend to copy the silence into another sample, you should use that sample's rate.

Seconds The number of seconds to be generated. Fractions are okay (e.g. 1.25)

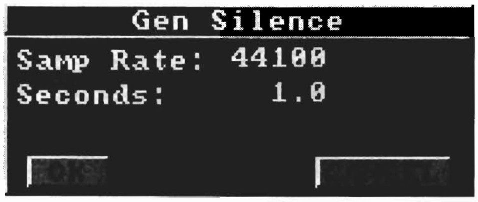


Figure 9. Generate Silence Selector.

Gen Sine Wave This option will create a pure sine wave in the CopyBuffer. A requester will appear allowing you to select the length of the sine wave and the sampling rate. Once the sine wave is in the CopyBuffer, you can select destructive edit and paste the sine wave into your working samples. This option is well suited to creating a 1,000Hz sine wave that is often used in the beginning of an audio tape for reference purposes.

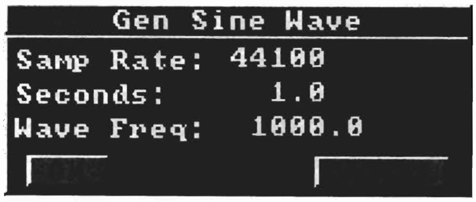


Figure 10. Generate Sine Wave Selector.

Sampling Rate The default sampling rate is from your working sample. If you intend to copy the sine wave into another sample, you should use that sample's rate. Note your Sampling Rate should be at least twice, preferably four times, the frequency of the sine wave.

Seconds The duration of the sine wave. Note that the longer sine waves the longer it will take to process.

Wave Frequency The frequency of the wave should be no more than 50% the sampling rate and preferably no more than 25%.

Menu: **Options**

Set range Using this option, you can enter a range numerically. Instead of dragging a range with the mouse, you can use **Set Range** to set the range exactly in either number of frames or number of samples (set by the Units in Samples menu option).

Set Start This is an "on the fly" operation that allows you to set a range while a sample is playing. Select Play All for a sample and then use the keyboard shortcuts for Set Start (*A*-M) and then Set End (*A*-E).

Set End Used in conjunction with set start allows you to set ranges when listening to a sample play. *A*-E.

Set Sample Parms

Use this to alter the sample's default volume, playback rate, as well as the filter rate for AD1012s, and the pan for AD516s.

The default volume is the level at which the sample plays. It will usually be +00dBs. The Mixer will automatically adjust itself to this level when the sample is played. However in Transport, if **Manual Vol** is selected, the current mixer setting will be used. Use Auto Volume to use the rate set here.

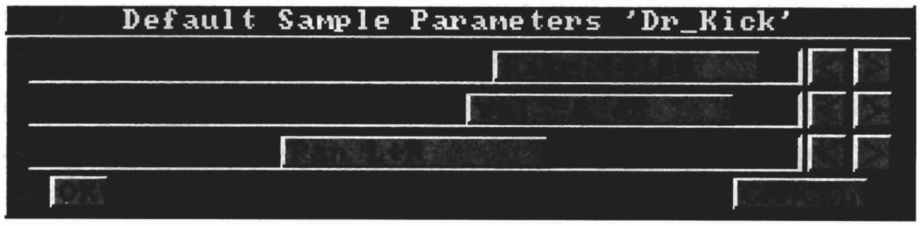


Figure 11. Sample Parameters Selector.

See the Reference Section on Recorder for explanation of sampling and filtering rates. If you have trouble selecting a sampling rate that is available in the Recorder, use the arrow buttons or click in the slider to the left or right of the knob. This will display all available increments. Just dragging the knob may not show them all.

The pan adjustment allows you to vary the level of audio to playing on each channel. The AD516 has two output channels, and you can set a sample to play on either the left, right, or both output jacks. When the pan slider is in the center, the sample will play equally on both the left and right output. If the pan slider is all they way to the right, the sound will only play on the right output. If the slider is all the way to the left, the sample will play only on the left. You can also vary the levels on each output.

Stereo is accomplished by recording two tracks: one panned full left, and the other panned full right. You use pans that are not full left or full right when playing back mono tracks that you want to appear on both the left and right outputs.

Set Display Offset

Prompts you for a time to offset either the display or the marked range. Changing either the offset will effect the other. If you know the exact frame you want the sample to trigger on, set the display offset to this frame. On the other hand, if there is an embedded sound effect in the middle of the sample (marked by a range), and it is crucial that it play on a specific frame, set the range offset to that frame number. The display offset will automatically calculate a new offset, to match the new range entry.

Loop Activate Loop to have the sample or marked range play continuously.

Freehand Draw

When activated, you can use your mouse to alter a sound's data. Selecting freehand will automatically zoom all the way in. To edit normally, deactivate the option or zoom out.

OK Requesters

When activated OK Requesters will appear for all destructive edits and effects. It is recommended that users new to Studio 16 leave "OK Requesters" activated until they become more familiar with Studio 16's operation. Advanced Users will probably want to turn off this option. A few crucial requesters are not optional and will remain even when OK Requesters is deactivated.

> NOTE: OK requesters can generally be accepted or rejected with the standard keyboard shortcuts Left *A*-V and Left *A*-B for OK and cancel.

Units in Samples

The graph and status display is generally displayed in SMPTE time code. If you prefer to work in number of samples, activate this option.

Show Regions

Regions are marked ranges that have been given a name. Regions are like samples in that they can be dragged into the Cue List and Transport. Show Regions will bring up an "Editor Regions List" that lists all the regions in a samples. You can drag from this region List. The "Editor Regions List" has Menu Options of it own.

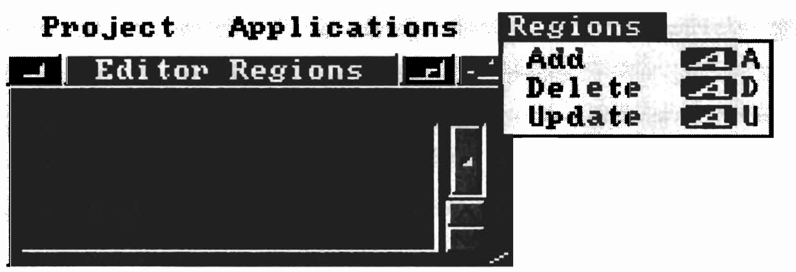


Figure 12. Editor Regions List and Regions Menu.

Add Region Brings up a name requester for the marked range. This range will then be converted to a region that can be dragged into the Cue List, or Transport. Its name will appear in the Editor Regions List along with any other regions existing for that sample.

Delete Region Highlight a region's name in the region list and select this option to forget the region. Note that no data will be lost, this is not a cut, or a non-destructive delete.

Update Region Allows you to adjust the region's length. Click on a region's name in the Editor Regions List, then click on the graph and move the start or end markers of the region. Next, make the Editor Regions List active by clicking on the title bar, then select Update Region to update the region with the new length.

Grid Displays a vertical grid that indicates boundaries within a sample. The type of grid is set with the submenu. For example, you can request that a vertical line be drawn on frame or one second boundaries.

Note that you may have to zoom in to see the grid lines. This is because the Editor will not draw grid lines when there would be "too many" drawn. In other words, if the editor thinks that you're going to have trouble seeing the waveform through all the grid lines, they will not be drawn.

Set the grid to one of the following settings, or turn it off. Note that the grid is a display only, it does not "snap".

OFF
1 Frame
1 Second
10 Seconds
1 Minute

Fast Graph When activated, the graph display update is quicker, but less representative of the actual sample.

Name: **Instance**

Keyboard: ^ I

Class: Application Module

Description: The Instance List can be used to launch Studio 16 modules just like the Applications Menu and Keyboard Equivalents. Instance is loaded from the Applications Menu. And once loaded, all other application or utility modules can always be on display rather than in the menu. Usually, due to limited screen space, it is preferable to use the Applications Menu and keyboard shortcuts.

Menu: **Instance**

Edit Brings up a requester giving you the option of renaming a module or making a module resident or non-resident. All Studio 16 modules reside on your disk in the Studio 16: Application, Drivers, and Utilities directories. For each module, there is one program file in this directory. As with all programs, a Studio 16 module must be in RAM to run. However, when you are not using a module, the module's code can be removed from RAM. The resident/non-resident option lets you decide if you want to keep a particular module's code in RAM even when the module is not being used (Resident) or whether you want to free RAM for use by other programs and remove a module when it is not is use (Non-Resident).

The advantage of keeping a module resident is speed. A resident module's window will open immediately when you double click its name in the Instance List since it doesn't have to be loaded from disk. A non-resident module will take a second or so to load when you double click its entry in the Instance List. For this delay you save a small amount of memory. The amount of memory (RAM) saved is relatively small so it's advisable to keep commonly used modules resident.

The Utility and Driver modules should always be resident. Application modules may be made resident at your discretion.

Duplicate This option creates another Instance of a particular type of module. For example, select the **Meters** entry and click the **Duplicate** button. You now have two entries in the Instance List, Meters and Meters#2. This enables you to load two Meters modules simultaneously, allowing you to show eight meters at once. However, many Amigas will not have enough CPU time for this.

Remove This option removes an Instance from the list. For example, if you don't require that second meter module (added in above example), you can select Meters#2 and then click the **Remove** button. Once a module is completely removed from Instance, you will have to load ModList to add it back.

Show Sets Instance to display Applications, Utilities, or both. Utility Modules are used in the operation of Studio 16. They are not usually accessed by the user. However, you may be interested in selecting the AD516 or AD1012Handler. It will display statistics about your hardware.

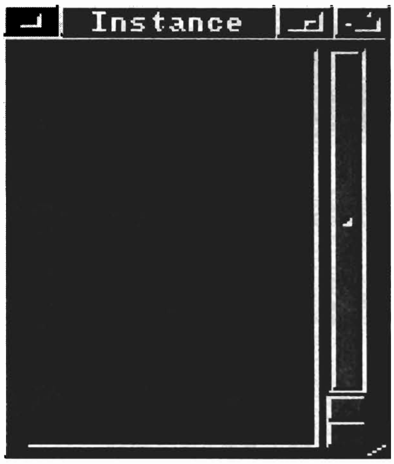


Figure 1. Instance List showing Application Modules

Name: **MessageMonitor**

Class: Application Module (Dormant)

Description: The Message Monitor is a handy debugging tool for Studio 16. This module is for programmers - most users will not require opening it. It displays all the messages passed between the different Studio 16 modules. It should be noted that message monitor can run at low task priorities, while most other Studio 16 modules run at higher priorities. This will cause a backlog of messages to be queued if the computer is fully loaded. These messages get processed by the higher priority Studio 16 modules before they are displayed by the message monitor.

Message Monitor is located in the dormant directory. This means that it is not available from the Applications Menu or the Instance List.

Procedure: To move Message Monitor from the Dormant directory:

1. Quit Studio 16
2. From WorkBench open the dormant directory, by clicking its icon.
3. Drag the Message Monitor icon on to the Application Directory Icon.
4. Delete the S:Studio16Instance.config file.
5. Load Studio 16, Message Monitor will appear in the Applications Menu and the Instance List.

Name: **Meters**

Keyboard: ^V

Class: Application Module

Description: This Module displays sound levels in much the same manner as the meters on a cassette deck. Like a cassette deck, these meters are used to adjust the input gain of incoming sound so you can get the maximum volume before clipping. Whenever you adjust the input gain of a card you should open a meter window. This allows you to make an intelligent decision about where the input gain slider should be set.

Another use of the meters is the visual monitoring of the sound going through each of the Studio 16 channels. You can easily tell if a sample was recorded improperly by looking at the level of a channel to see if it is too low or too high. You may decide that a sample needs to be re-recorded.

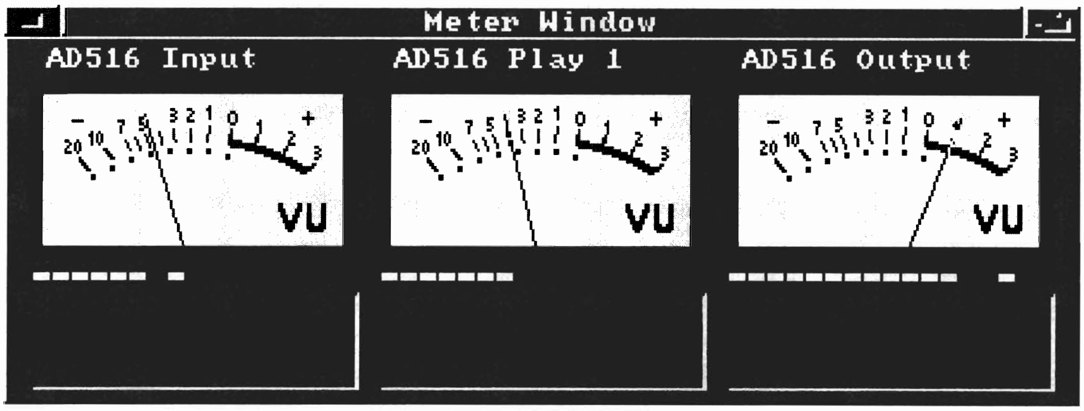


Figure 1. Meters

Layout: The meter window can consist of an Analog Meter, a Digital Meter and a scrolling graph. Each meter is assigned to a specific channel using the Channel Menu option. Each meter takes up more blitter time, so slower systems should keep the number of meters to a minimum.

For most applications, the gain information required can be derived from the digital meters and the scrolling graph. Other than setting the input gain, meters are optional. While they look nice, they also take up memory and a surprising amount of CPU and blitter time. It is quite possible that if you close down the Meter module you will be able to playback more channels.

Menu: **Meters** Allows you the flexibility to show only the types of meters you require. Each meter channel can display the volume information in up to three ways: an analog meter simulation, digital meter simulation with peak hold, and a scrolling graph.

Analog The analog meters use a waited averaging technique to show the average sample amplitude. Large instantaneous peaks in a sample's volume will have only a small effect on an analog meter's needle position unless they occur frequently. It is difficult to decide if a sample is clipping using this type of meter, so it is more useful to use one of the other types of meters for adjusting the input gain of the card.

Digital The digital meter shows the instantaneous peak of a sample's amplitude in a given period of time. If a sample has one large peak this type of meter will display the peak. If the right most LED is displayed, then clipping occurred. In most cases the input gain should be reduced to prevent this. The peak hold feature shows the largest amplitude of a sample for a longer period of time. This makes it easy to determine how loud the peaks of a sample are.

Graph The scrolling graph meter is a type of meter unlike any analog world counterpart. It shows the instantaneous peak of a sample on one axis while showing time on the other axis. This type of meter is useful for determining the type of sound being played through a channel as well as its amplitude. With practice you can tell whether a channel is playing back a voice track, a sound effect or some music just by looking at the shape of the graph.

Menu: **Channels** Allows you to assign meters to specific channels. Select the channels required. Input should be selected to monitor the gain level before recording.

> Note: To select multiple options in a sub-menu click the left mouse button while holding down the right.

The maximum number of meters per meter module is four. To show more than four meters, you must first close the Meter Window and launch Instance from the Applications Menu. From Instance duplicate Meters. Now from the Applications Menu you can access Meters#2, this will provide four additional meters. For more meters, duplicate Meters in Instance again.

Name: **Mixer**

Keyboard: ^ M

Class: Application Module

Description: The mixer allows you to adjust the relative volumes of the playback channels and the input and output volume. And, if the AD516 is installed, a pan adjustment is included for each playback channel. The mixer uses the DSP chip to digitally adjust the volume and pan of each channel. You should use the mixer to adjust a sample that is playing back too loud or too soft or to make a sample playback on the left or right channel (or both). The volume and pan may be changed before, after or during sample playback. The mixer also displays the sample name currently playing through each channel.

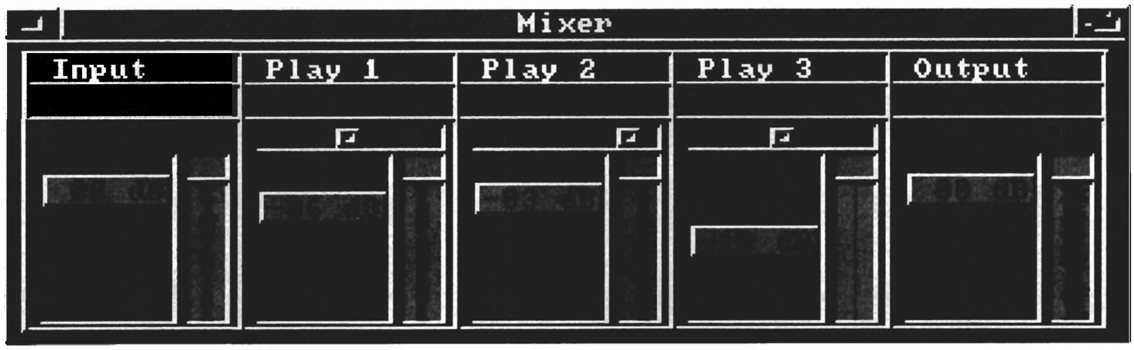


Figure 1. Mixer

Volume To adjust the volume of a channel simply move the appropriate slider. The mixer will display which sample is playing on which track. This makes it easy to identify which volume slider to adjust. Use the output volume slider to adjust the volume of all the channels. This is useful for fading in or out entire soundtracks. The input volume is useful when you want to mix the card's playback channels with an external audio source.

Since the mixer works in the digital realm it is almost distortion free; however, care should be taken to refrain from clipping a channel since clipping causes distortion. Digital distortion is somewhat harsher sounding than analog distortion.

Note: Ideally, a channel should be maintained at +00dB where the dynamic range is maximized. When you reduce a channel's volume digitally (as the mixer does), you are also reducing the channel's dynamic range.

When recording from the Recorder or from the Transport Module in the Record Input Mode, the audio input will be digitized regardless of the Mixer setting. Note on Figure 2, the Record Input Tap is before the volume adjustment on the Mixer. To record at the output tap. You must use Transport and set the options to Record Output.

Decibels The Mixer specifies volumes in Decibels (dBs). Decibels are a logarithmic scale used to more accurately represent how the ear hears volume changes. Zero dB means no gain. Attenuation are negative, and amplifications are positive.

For example:

-∞dB is equivalent to no volume
-6dB is equivalent to 50% volume
0dB is equivalent to 100% volume
+6dB is equivalent to 200% volume

-∞, or negative infinity is represented by -00dB in Studio 16.

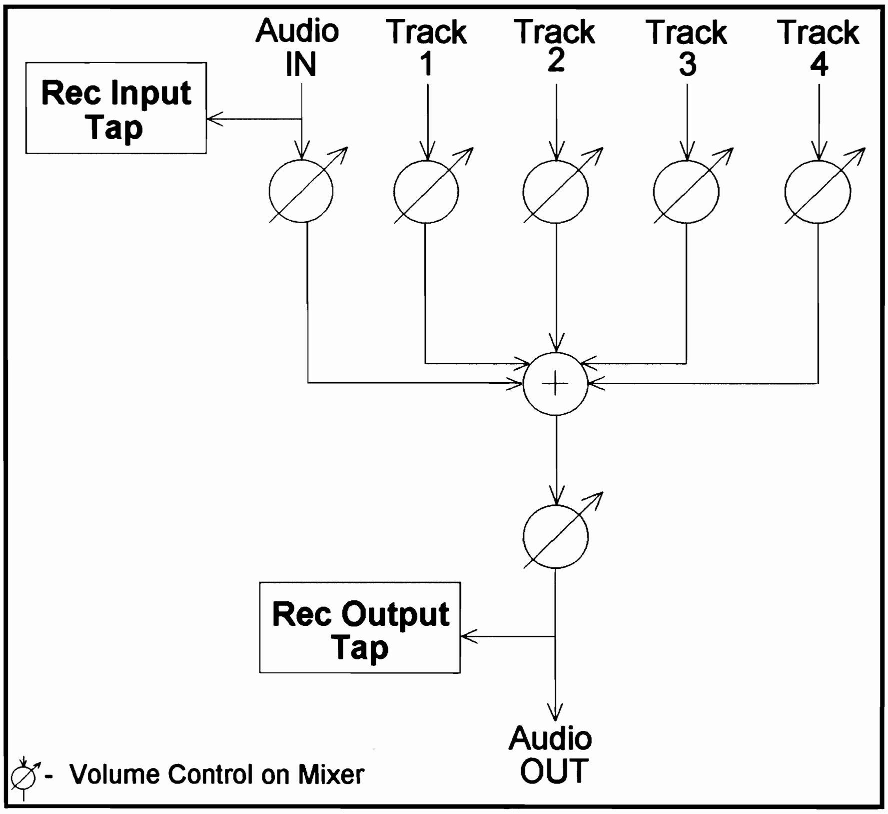


Figure 2. Mixer Diagram

Pan The pan adjustment allows you to vary the level of audio to playing on each channel. The AD516 has two channels, and you can set a sample to play on either the left, right, or both output jacks. When the pan slider is in the center, the sample will play equally on both the left and right output. If the pan slider is all they way to the right, the sound will only play on the right output. If the slider is all the way to the left, the sample will play only on the left. You can also vary the levels on each output.

Stereo is accomplished by recording two tracks: one panned full left, and the other panned full right. You use pans that are not full left or full right when playing back mono tracks that you want to appear on both the left and right outputs.

NOTE: The AD1012 is a mono card and does not have a pan option.

Clip Indicator

The clip indicator is similar to the peak hold on the digital meter. It indicates when the channel's volume clips.

Scrolling Graph

Similar to the Scrolling Graph in Meters. It is also possible that the mixer can load down the system. This is true for any of the windows that responds to real time events. If the system is too busy, Studio 16 may be limited in the number of simultaneously playing channels. Closing the mixer window may increase the number of channels you can play back. Or, you can try substituting Tiny Mixer instead.

Menu: **Mixer** Allows you to allocate a specific number of channels for the Mixer. Select the channels required. Input should be selected to monitor incoming audio.

Note: To select multiple options in a sub-menu click the left mouse button while holding down the right.

Name: **ModList**

Class: Application Module (Dormant)

Description: The ModList, lets you load or unload code from disk and memory. If you have modules that are not in your Instance list, but are on disk in your Application, Utilities, or Drivers directories, you can use ModList to load the module. After you have loaded the module it will be listed in Instance. You can load a module multiple times to create multiple instances.

> NOTE: The **ModList** will not be used by most users. It is included for advanced users or for those rare occasions that require its use.

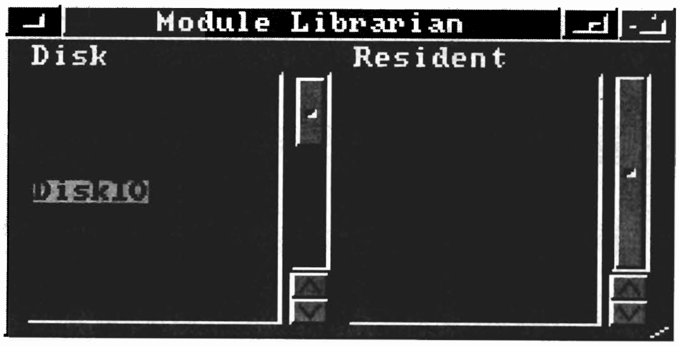


Figure 1. ModList (Module Librarian)

Layout: In the ModList, the left list contains modules on disk in your Applications, Utilities, or Drivers directories, see Figure 1. The right list contains modules that are currently in RAM. Double clicking an entry in the left list (the "on disk" list) will cause the module to load into RAM. Generally the loaded window won't open unless you select it in the Instance List. Double clicking an entry in the right list (the "in RAM" or "Resident" list) will cause the module to unload itself. However, its "instance" will remain in the Instance List until removed.

Procedure:

Move ModList from the Dormant directory

1. Quit Studio 16.
2. From WorkBench open the dormant directory, by clicking its icon.
3. Drag the ModList icon on to the Application Directory Icon.
4. Delete the S:Studio16Instance.config file.
5. Load Studio 16. ModList will appear in the Applications Menu and the Instance List.

Name: **Open List**

Keyboard: ^ O

Class: Application Module

Description: The Open List contains lists of your samples by directory. You can play, rename, delete, and edit, from Open List, Figure 1. Launch the Open List by selecting it from the Applications Menu, or type ^ O.

A sample in the Open List is a digitized sound on your hard disk, that you have recorded or loaded. Since Studio 16 always works with files on the hard disk, it is a little different from most programs. For example, there is no need to save each sound before you quit. If you quit Studio 16, then re-run it later, all your "open" sounds in the open list will still be there. Of course, if you have your samples in RAM:, your samples will be lost when you turn off your Amiga unless you save them to disk.

All the samples in the Open List are draggable to the Cue List and Transport. As are the regions that can be displayed in Open List, (activate Show Regions).

All the Open List Menu options can be executed on multiple samples. Select multiple samples by holding down the shift key, and clicking on the Samples' names.

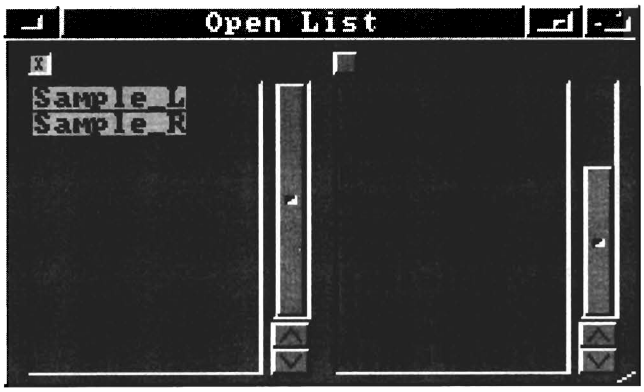


Figure 1. Open List

Gadget:

Record Path

The record path is the directory where samples will be recorded to. It is indicated by an active box (☒)next to the Directory name. To change the record path, add another path and then click the active box next to the new directory's name.

Menu: **Open List**

Play Sample

Select this option to play a sample. Select multiple samples by shift clicking. When playing back multiple samples, the sampling rate must be the same for all samples. The AD516 is limited to 8 simultaneous samples, the AD1012 is limited to 4.

AD516 ONLY - A stereo sample will appear in the Open List as two samples, like: Untitled_L, and Untitled_R. Play both channels by selecting both samples in Open List. Shift clicking will enable you to select both of them, hit *A*-P to play.

To adjust the volume and pan of samples while they play, load the Mixer from the Applications Menu and make level adjustments in realtime.

Stop Playback

To stop playing sounds, select the Stop Playback option.

Rename Sample

The **Rename** option allows you to change the name of samples in Open List. Select one or more sample names so that they are highlighted. Select the **Rename** Samples option. For each sample selected, a requester will appear asking you to type in the new name for the sample.

Delete Sample

To remove a sound permanently from your hard disk and the Open List, select its name in the Open List then select the Delete option. This will delete the file.
CAUTION: There is no way to recover a deleted sample.

Convert Sample

This option allows you to convert the file format of a sample to an alternate format. Once a Studio16_2.0 sample has been converted to another format, it can still be accessed by Studio 16, but any region parameters and/or SMPTE start times that were associated with it may be lost.

When converting files to eight bit formats, you will decrease the size of the file size by half, and you will also lower its fidelity. However, if you are working with samples that don't require a high Signal-to-Noise Ratio, (e.g. explosions, crashes. etc.) converting them to an 8 bit format will save disk space and still allow you to play them simultaneously with your 16 bit samples, assuming they were recorded at the same rate.

Available formats are:

Studio 16_2.0 - Studio16_2.0 is a 16 bit format, similar to AIFF, but it appends special data that keeps track of non-destructive edits and regions. Note that even though the AD1012 records with 12 bits of resolution, it stores files using 16 bits.

Studio16_1.0 - The original Studio 16 format.

AIFF 16 bit - This is a standard file format for 16 bit files. Studio 16 edits, loads, and saves sounds with 16 bits. AIFF is very common on the Macintosh, and is used by some Amiga software. AIFF does not remember regions or non-destructive edits.

AIFF 8 bit - This is an AIFF format that only uses 8 bits. If you convert to this format, you will lose some sound quality since you are dropping bits. However, the file's size will be reduced by about half.

IFF 8SVX - This is a very common Amiga format for 8 bit samples. Files convert to this format can be loaded into 8 bit sound editors, such as Perfect Sound or Audition 4. If you convert to this format, you will lose some sound quality since you are dropping bits. However, if the file is a sound effect you may not notice the decrease in fidelity, and the file's size will be reduced by about half.

RAW - This stores raw sample data in two's complement binary form. The first sample will be in the first word of the file (each sample is 16 bits), the second sample will be in the second word (third and fourth bytes), etc. This format is not generally recommended, except perhaps for use by programmers who want a simple file format to load.

CDTV RAW - This format is only of use to developers with the CDTV emulator. This format will create a stereo CDTV raw audio file by duplicating mono data on the left and right channels.

Edit Sample

Selecting a sample name then the **Edit** option will bring up the sample in an editor. See the **Edit** Reference Section for a more complete description.

Add New Path

Select this option to add another directory to the Open List. This is especially useful if you are working with multiple hard drives or you have samples organized into multiple directories. When selected, a path requester will appear allowing you to select a new path. Click OK to update Open List. A maximum of eight directories can be listed at once.

Remove Path

Select Remove to bring up a Remove Path requester.

Update Path

Selecting this option will update the current directories in the Open List. This is especially useful if you are using a removable media drive like SyQuest, and you change cartridges, or if you are copying files from DF0 and you change disks.

Show Regions

When activated, a region list will be added to the right of your paths. The region list will display the regions that have been selected for a specific sample. Select a sample with regions, and they will be displayed in the region list.

Name: **Preferences**

Keyboard: ^P

Class: Application Module

Description: Preferences is located in the Project Menu. It is dedicated to customizing the appearance and performance of Studio 16. Included in Preferences are: Interface Options, SMPTE source, Memory Options, and the default Record Format.

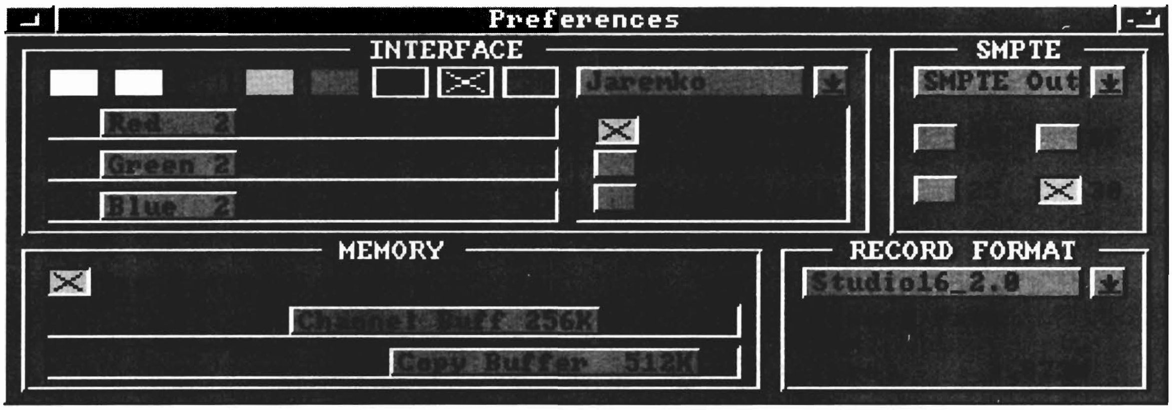


Figure 1. Preferences

Layout: The Preferences window is divided into four sections: Interface, SMPTE, Memory and Record Format.

Gadget: **Interface -**

Color Boxes and Sliders
Click on a Color Box to select which color to adjust. Slide the knob on the red, green and blue sliders to adjust the color. The affected areas will update as you adjust the slider. Click Save Setup in the Project Menu to have Studio 16 use the new colors on loading.

Preset Drop List Click on the ↓ to view a list of preset screen colors. Use the scroll bar and arrow buttons to scroll through the options. Click a selection to make the Studio 16 screen update using the new colors.

Interlace By selecting Interlace, your screen will double in vertical resolution resulting in a larger work area. Running an Amiga 2000 in Interlace mode without a flicker fixer may result in an undesirable flicker.

To activate the Interlace mode, you must select Interlace, select SaveSetup from the Project Menu, Quit Studio 16, and then restart the program.

ClickToFront Activating this button will allow you to bring windows to the front by simply clicking anywhere on them. If it is not selected, you bring a window to the front by clicking the Depth Gadget in the title bar of windows, or by calling them to the front with the keyboard shortcut.

SimpleTitleBar This button will cause the Close, Minimize and Depth button to only show in the active window. All non-active windows will only display the module name.

Gadget: **SMPTE -**

SMPTE Source Click on the droplist to select your default SMPTE source. The recommended setting is either AD516#1 or AD1012#1. This will set your default Source to your External SMPTE; however, it will be overridden to SMPTE gen when you open SMPTE Generator. Closing SMPTE Generator will cause Studio 16 to revert to your default setting, (e.g. AD516#1).

SMPTE Rate By selecting 24, 25, or 30 Frames per Second, or DF, you select which SMPTE Time code format to be used by the SMPTE Monitor and Generator, Sample Editor and Cue List Module. DF is 30 fps, drop frame. '30" is non-drop frame.

The common applications for the available formats are:

24	Motion Pictures
25	European Video
30*	USA B&W Video and Music
DF	USA B&W Video and Music

*If you are using 29.97 non-drop frame, select 30 fps.

For more information on the various SMPTE formats see Chapter 5 - SMPTE.

Gadget: **Memory -**

Use Extended Memory

If selected, this button causes Studio 16 to try to allocated channel buffers in extended memory first, followed by AutoConfig fast RAM, and lastly chip RAM. If it is NOT selected, this will prohibit Studio 16 from allocating channel buffers in extended memory. This button only effects channel buffers.

You **SHOULD** select this button if your hard disk controller and '030/'040 card are on the same PCB. (e.g. You have a stock A3000, or You have a GVP 030 card with built in hard disk controller.)

You **SHOULD NOT** select this button if you have an '030 or '040, but your hard disk is on a stand alone Autoconfig card. (e.g. You have a GVP series II DMA hard disk controller and a PP&S 040 card with 32 bit RAM.)

Your Amiga has two major types of memory: Chip and Fast. Chip memory is directly accessible by the Amiga's graphics chips. Fast memory is not directly accessible by graphics chips, but can be accessed faster than Chip memory.

And, there are roughly two different kinds of Fast RAM:

Extended Memory is fast RAM that is only accessible by the 32 bit 68030 or 68040 processor. It lies in address $FF000000 to $FFFFFFFF.

Zorro II AutoConfig Memory is fast RAM that can be accessed by a standard Zorro II AutoConfig card. It is also referred to as "24 bit" or simply "Autoconfig" memory.

Studio 16 allocates "Channel Buffers" in RAM when playing a sound. Data is buffered there on its way to or from the hard disk. For efficient data transfer, DMA hard disk controllers need direct access to these buffers. This means that if you have a Zorro II AutoConfig DMA hard disk controller, your channel buffers can not reside in extended memory. This is because AutoConfig cards can not DMA into extended memory.

Channel Buf The Channel Buffer specifies how much RAM you want to use for buffering each sound that plays back. When playing back multiple sounds simultaneously, a channel buffer is allocated for each playing sound.

If you have less than 3 MB of RAM, reduce the default channel buffer size from 256K to about 64K. You may want to increase the default channel buffer size if you have a slow hard disk, or you encounter "skipping" while playing sounds. See the Chapter 7 - Troubleshooting for more information on skipping.

Keep in mind that one channel buffer is allocated for each playing sample. Make sure you have enough system RAM to cover your requested buffers. The chart below lists the RAM used by a specific number of playback samples. For example, if you plan to play four simultaneous tracks with the channel buffer set to 4096K, you will need 16,384K (or 16 MB) of free RAM just for buffers.

This same chart will calculate the RAM used by Cue List for a specific number of preloads. The number of samples to preload is set from the Prefs in the Cue List Menu.

The default setting (256K) is adequate for most systems.

Channel Buf Size	RAM Usage by a Specific Number of Tracks Playing Simultaneously							
	1	2	3	4	5	6	7	8
128	0.1MB	0.3MB	0.4MB	0.5MB	0.6MB	0.8MB	0.9MB	1.0MB
256	0.3MB	0.5MB	0.8MB	1.0MB	1.3MB	1.5MB	1.8MB	2.0MB
512	0.5MB	1.0MB	1.5MB	2.0MB	2.5MB	3.0MB	3.5MB	4.0MB
1,024	1.0MB	2.0MB	3.0MB	4.0MB	5.0MB	6.0MB	7.0MB	8.0MB
2,048	2.0MB	4.0MB	6.0MB	8.0MB	10.0MB	12.0MB	14.0MB	16.0MB
4,096	4.0MB	8.0MB	12.0MB	16.0MB	20.0MB	24.0MB	28.0MB	32.0MB

Figure 1. Channel Buffer Size required for Number of Simultaneous Tracks

Copy Buffer Size The Copy Buffer specifies how much memory to use in many Edit, Save, and Load operations. The larger the CopyBuffer, the quicker these operations will occur. The default is 32K. (This setting is not related to the CopyBuffer sample created during destructive edits.)

Gadget:

Record Format -

When a sample is recorded it is usually stored in the Studio16_2.0 file format. This default format is selected in Preferences. If your file is to be exported to another sound program you can convert it to another file format in Open List. You can also record it direct to disk in that format.

When converting files to eight bit formats, you will decrease the size of the file size by half, and you will also lower its fidelity. However, if you are working with samples that don't require a high Signal-to-Noise Ratio, (e.g. explosions, crashes. etc.) converting them to an 8 bit format will save disk space and still allow you to play them simultaneously with your 16 bit samples, assuming they were recorded at the same rate.

Available file formats are:

Studio 16_2.0 Studio16_2.0 is a 16 bit format, similar to AIFF, but it appends special data that keeps track of non-destructive edits and regions. Note that even though the AD1012 records with 12 bits of resolution, it stores files using 16 bits.

Studio16_1.0 The original Studio 16 format, similar to Studio16_2.0 but will not support regions.

AIFF 16 bit This is a standard file format for 16 bit files. Studio 16 edits, loads, and saves sounds with 16 bit files. AIFF is very common on the Macintosh, and is used by some Amiga software.

AIFF 8 bit This is an AIFF format that only uses 8 bits. If you record in this format, you will lose some sound quality since you are dropping bits. However, the file's size will be about 50% smaller.

IFF 8SVX This is a very common Amiga 8 bit digital sound format. Files recorded in this format can be loaded into 8 bit sound editors, such as Perfect Sound or Audition 4. If you convert to this format, you will lose some sound quality since you are dropping bits. However, if the file is a sound effect you may not notice the decrease in fidelity, and the file's size will be reduced by about half.

RAW This stores raw sample data in two's complement binary form. The first sample will be in the first word of the file (each sample is 16 bits), the second sample will be in the second word (third and fourth bytes), etc. This format is not generally recommended, except perhaps for use by programmers who want a simple file format to load.

CDTV RAW This format is only of use to developers with the CDTV emulator. This format will create a stereo CDTV raw audio file by duplicating mono data on the left and right channels.

Record Path - Is selected in the Open List. A display is shown in Preferences.

Free - The displayed number next to Free: is the number of available megabytes in the selected record path.

Name: **Quit**

Keyboard: ^Q

Class: Project Module

Description: Select Quit, from the Project Menu when you want to end your Studio 16 session. A requester will appear to verify you want to quit. To save the current window arrangement and Preference settings, select Save Setup from the Project Menu before quitting.

> NOTE: If you have recorded into RAM, be sure to move your samples to hard disk before turning off your Amiga.

Name: **RealtimeDelay**

Keyboard: ^D

Class: Application Module

Description: The Realtime Delay allows Studio 16 users to process audio in real time using the DSP on board. The modulation delay combines the controls of a standard delay with delay modulation parameters more commonly found on effects boxes such as flangers. These controls allow you to create numerous effects including delays, choruses and flanges in realtime. Realtime, refers to audio as it plays through the AD516 or AD1012, not samples playing from hard disk.

NOTE: This module takes over Studio 16. This means that no other modules will work and no other audio may be played back while the card is in this mode.

Several different effects can be obtained by altering the parameters of the modulation delay. The two most common effects are delays, and flanges.

Delay: Delays and echoes are achieved by having a delay time of at least 30ms and zero modulation. Slapback echo can be obtained by moving the Feedback slider to zero percent, while normal echoes are accomplished by increasing the feedback. Decrease the Delay Level control if the echoes are drowning out the original signal.

Flange: This effect occurs when the time delay is below 20ms. Short time delays cause phase cancellation at certain frequencies. If you change the time delay, you also change the cancellation frequencies. This effect usually uses modulation to produce a sweeping effect. To produce this effect use a fast sampling rate and small time delays. Also your Delay Level should be nearly set at full volume. The effect can be enhanced by increasing the feedback or for more subtle effects reducing feedback.

Layout: The window is divided into three main parameter sections; System, Delay, and Modulation. Figure 1.

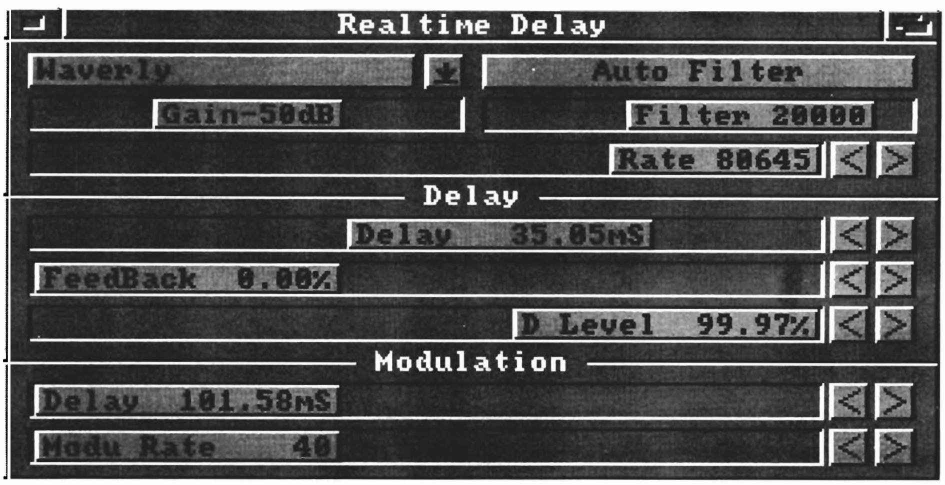


Figure 1. RealtimeDelay

System Parameters -

Defaults This droplist contains some preset parameter combinations that illustrate possible effects. The Dry setting is the sample playing though without a delay or modulation.

AutoFilter Available only on the AD1012, this selects an appropriate filter setting for the selected Sampling Rate. Refer to the Recorder Reference Section for details.

Gain Adjusts the input gain of the AD516 or AD1012. Refer to the Recorder Reference Section.

Filter Adjusts the cutoff frequency of the anti-aliasing filter. Refer to the Recorder Reference Section.

Rate Adjusts the sampling rate of the card, it also effects the delay time. Slower sampling rates produce longer delays than faster sampling rates. For flanges you will probably use a faster rate. Refer to the Recorder Reference Section.

Delay Parameters -

Delay Adjusts the length of time between repeats. This parameter is also used as the upper delay time during modulation effects.

Feedback Controls the number of echoes you hear. This is accomplished by adding a percentage of the delayed signal back into the delay.

D Level Controls the volume of the delays. At 100%, the first delay and the original signal are the same volume. At 0%, you will not hear a delay, resulting in a "dry" sound.

Modulation Parameters -

Delay Controls the lower boundary of a modulation delay. If this value is the same as the first delay parameter, you will hear a normal delay. But if the value differs, the delay time will increase and decrease between the two boundaries, creating a modulation delay.

Modu Rate The Modulation Rate controls the rate of increase and decrease between the two delay boundaries.

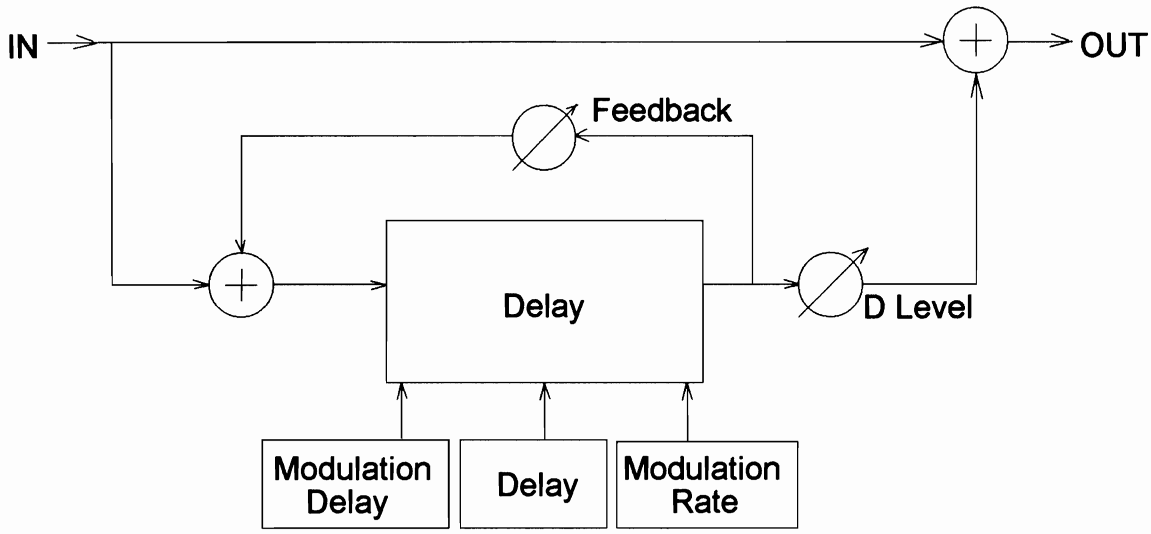


Figure 2. Delay Diagram

Name: **Recorder**

Keyboard: ^ R

Class: Application Module

Description: The Recorder Module allows you to monitor incoming audio, adjust the gain, filter and sampling rate, and record audio to hard disk.

The Meter Module is often used in conjunction with record. By evaluating the input levels on the meters, the gain level is easier to adjust.

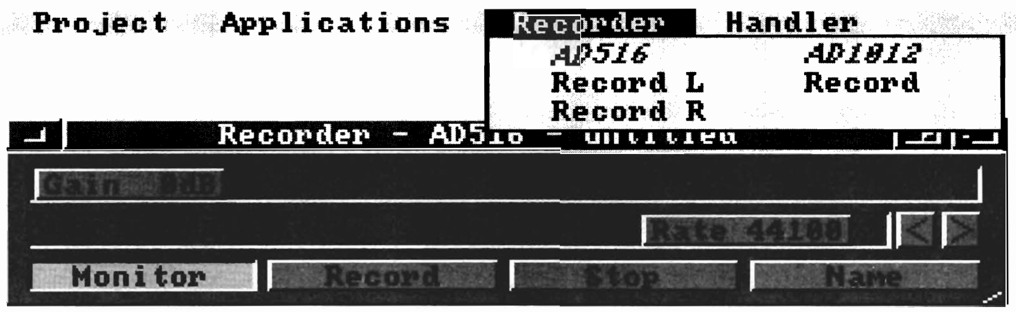


Figure 1. AD516 Recorder and Recorder Menu

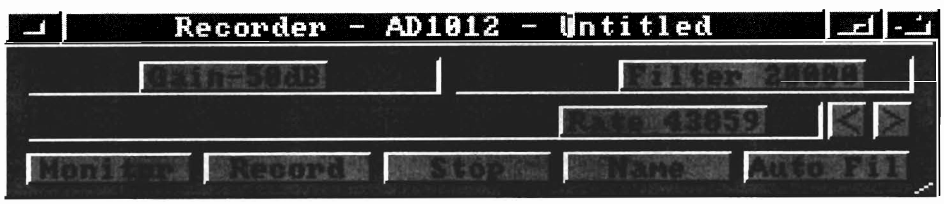


Figure 2. AD1012 Recorder

Procedure:

Record a sound with the AD516 or AD1012

1. Set up your audio connections as mentioned in the Installation Chapter.

2. Launch Recorder, Mixer and Meter from the Applications Menu, ^R^M^V.

AD516 ONLY - From the Recorder Menu, select Channel L and Channel R to record a stereo sample. The AD1012 will record a mono sample.

3. Click **Monitor** in the Recorder.

4. With **Monitor** activated, you will be monitoring the audio input. This means the audio entering the "audio in" jacks is being digitized and passed back out the "audio out" jacks. The sound is being digitized, but not stored on your hard disk.

5. With monitor on, you should be able to hear your audio source, and you should see activity in the Mixer and Meters. If this doesn't occur, check your audio sources and connections. Check the Mixer input and output level, they should both be at -+0dB.

   The Mixer has a scrolling graph for each channel. When Monitor is turned on, you will see activity in the Mixer's input and output channel.

AD516 ONLY -	If you are recording in stereo. The scrolling graph will be symmetrical, the recorder is digitizing both the left and right channels. Unplug the audio from the left channel to see the result of a mono sample on the scrolling graph.

6. Once you are hearing audio, and see activity in Mixer and Meters you can adjust the **Rate**, **Filter**, and **Gain** controls. (A detailed description follows in gadget reference.)

7. Click the **Name** button to assign a name for the sample you are about to record. When prompted, type in the name.

8. When you are ready to record, click the **Record** button. A Record Status Display will appear with the size of the sample being recorded and the remaining disk space available in the record path.

9. Click the **Stop** button to stop recording.

10. Check Open List for the new sample.

11. Play the sample by selecting the sample's name in Open List and hitting *A*-P.

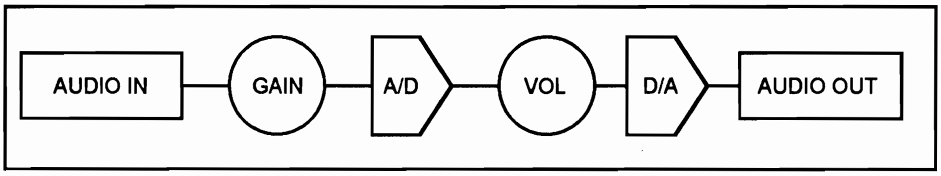
AD516 ONLY -	A stereo sample will appear in the Open List as two samples, like: Untitled_L, and Untitled_R. Play both channels by selecting both samples in Open List. Shift clicking will enable you to select both of them, hit *A*-P to play.

Gadget :

Gain The gain slider adjusts the gain of the card's internal input amplifier. This is similar to a "record level" on a tape deck. You generally want to keep the gain as high as possible without causing "clipping". Clipping occurs when the input signal exceeds the analog to digital converter's range. You can see clipping on your input meters and Mixer. In the Meters, it shows up in two ways: the digital VU meter in the meter window will hit the far right side and turn another color at the end. You can also see clipping when the scrolling waveform touches the top and bottom of its box often. In the Mixer, the peak indicator over the scrolling graph will flash. Clipping causes distortion in your recorded audio and should be avoided by reducing the gain.

The gain level required varies from device to device. Also, when adjusting the gain, you may hear small, rapid clicks. This is called "zipper noise" and is normal on the AD1012. On the AD516, monitoring will stop while you are adjusting gain.

There is a big difference between GAIN and VOLUME in Studio 16. The gain setting adjusts the gain on analog amplifiers before the Analog to Digital converter. Volume settings (as in the mixer) use the DSP to digitally attenuate or amplify a track. When recording a new sample, you should set the Mixer volume sliders to +0dB, and adjust the gain for full scale. You can always lower the volume on playback., but on record for maximum Signal-to-Noise Ratio, you should always record at full scale.



If adjusting the gain level has no effect on the audio level, refer to Chapter 7 - Troubleshooting.

Filter This feature is only available on the AD1012. It sets the -3dB cutoff point of the 8th order low pass anti-aliasing filters of AD1012. There are two filters: one on the input and one on the output. Both are always set to the same frequency. (The AD516 has a much sharper digital filter. It's always set to .45 times the sampling rate.)

Preferably, this filter should be set to half the sampling rate, but you can vary it for different effects.

Rate This sets the sampling rate for use by **Monitor** and **Rec**. For example, 44100 Hz (44.1 KHz) is the sampling rate used by CD players and gives a theoretical 22KHz frequency response. The theoretical frequency response of digitized audio is half the sampling rate.

Studio 16 can record sound at many different sampling rates. The faster the sampling rate, the better the frequency response of the recorded signal. Unfortunately, with faster sampling rates, more disk space is used. And just as importantly, faster sampling rates mean faster data transfer rates. As a general rule of thumb, with a 44KHz sampling rate, you will use 5MB per minute of audio per channel. See the space requirement chart in Chapter 4.

As you move the rate slider, the sampling rate will change to a new value. The AD516 and AD1012 only have certain distinct rates available--set by the card's master clock. In other words, the cards can not sample at any arbitrary rate. Instead, you must select a rate from the predetermined rates available. (For the AD1012 these rates are determined by the formula: RATE = 10,000,000÷ x, where x can range from 122 to 1320. The number 10,000,000 is from the DSP's 10MHz master clock.) The **Rate** slider will show you all available rates.

Users often feel the need to use a 44K sampling rate since it is the CD standard. But, if your final distribution media isn't a CD, there's not much point in using that fast of a sampling rate. Often, a 30K-32K sampling rate is just as good for material that will end up on tape. While monitor is on, try adjusting the rate slider and listening to the effect it has on audio quality. Pick the lowest rate that gives you acceptable audio quality. Once you pick a rate you should try to record all your samples at the same rate. This will prevent any problems when mixing samples.

There are a few situations that may require you to use lower sampling rates:

- Your hard disk is full
- Your hard disk is skipping during playback
- Your screen flashes

See Chapter 7 - Troubleshooting - for more information.

Monitor Click this button to begin monitoring the Audio going into the A516 or AD1012. To adjust the level of the **Monitor**, you can load a Mixer and adjust the monitor or input channel by sliding the associated bar up or down.

Record Click this button to start recording.

Stop The Stop button is clicked to stop recording. The hard disk light may continue to flash for 1-20 seconds. This is normal and is dependent on your channel buffer size and hard disk speed.

Auto Filter Only available on the AD1012. Activating Auto Filter will cause the Filter Value to "track" the Sampling Rate at the preferred relationship of: Filter Rate = ½ Sampling Rate. Changes in the Sampling Rate will automatically cause the Filter rate to update as long as auto filter is activated.

Name This button must be selected to name a sample before making a recording. The default setting is Untitled. If the sample name has already been used, Studio 16 will append a number to the new recording. To change a name of a sample after is has been recorded, you can use the Rename option in the Open List Menu.

NOTE: Do not use a greater than symbol (>) in a sample name. It is reserved to delineate regions.

Menu: **Recorder**

AD516 ONLY When the AD516 is installed, the Recorder Menu allows you to select the channel(s) to record. If you are recording in mono, select either Channel - L or Channel - R. For stereo, activate both channels.

If you have an AD1012 and an AD516, you can record all 3 channels simultaneously, however, the exact synchronization of the AD1012 is not guaranteed.

Menu: **Handler** This menu only appears if you have more than one AD516 or AD1012 installed, it allows you to selects which card the recorder settings will affect.

Name: **SMPTE Generator**

Keyboard: ^ G

Class: Application Module

Description: The SMPTE Generator module allows you to generate internal SMPTE time code, Figure 1. You can use the internal time code generator to drive the Cue List and the SMPTE Monitor modules or use it to sync Studio 16 with Bars&Pipes Pro.

When SMPTE Generator is opened, it automatically selects itself as the current SMPTE source. When you close the SMPTE generator, the default source becomes active again.

The main advantage of the internal time code generator is its ability to trigger Cue Lists. Anyone can set up long audio sequences of different samples without using the editor to do destructive editing. The Cue List allows you to specify the starting time code and mix level of an unlimited number of sounds.

The SMPTE Generator does not output physical time code, or enable you to stripe tapes.

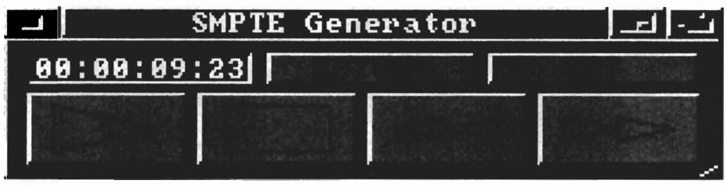


Figure 1. SMPTE Generator

Gadget:

Return This button will stop time code generation and reset the time to 00:00:00:00, or the most recent set time.

Set Allows you to program the return frame number. You do not have to type in colons (:). Type 300, for the reset time be to 00:00:03:00. After it's programmed, hit the return button to update with the new time. The recommended set time is 1 second before the first sample triggers. This allows time for samples to preload.

Play This will start time code generation from the current value displayed in the upper left corner. If Cue List has entries listed, and it's turned on, clicking play here will trigger the Cue List.

Stop This will stop (pause) the generation of time code, to continue generation click (play).

Rewind The first click will advance the counter at normal speed in the reverse direction. Each additional click will increase the speed. Time code is not generated in this mode.

Fast Forward The first click will advance the counter at 5 × normal speed. Each additional click will double the speed. Time code is not generated in this mode.

Name: **SMPTE Monitor**

Keyboard: ^ S

Class: Application Module

Description: The SMPTE Monitor allows you to view incoming external or internal SMPTE time code. It is useful for viewing the value of external time code coming into a card, Figure 1. This module can allow you to do pencil edits without having to record a SMPTE window box on video tape. If you own a genlock, you can use this module to produce a SMPTE window dub. Select Hide Title Bar from the SMPTE Monitor Menu.

Figure 1. SMPTE Monitor

The SMPTE monitor displays the current time code in its window. The SMPTE monitor also looks for three types of SMPTE errors, invalid time code and SMPTE time-out and SMPTE jump. If a time code is deemed invalid by Studio 16, one of three small squares in the upper left corner of the SMPTE Monitor window will flash. It is normal for this square to flash occasionally, such as when you start and stop time code. If the square flashes constantly, check to make sure that you have set the correct SMPTE frame rate in Preferences.

From Left to Right, the squares mean:

SMPTE Time Out No time code was detected for about 1/2 a second. Pausing your video deck will cause this.

SMPTE Time Code Error This error is generated if an illegal time code is read by the SMPTE reader. For example, a time code with 69 seconds would generate this error. This error typically results from distortion, usually a bad cable, a bad connection, or tape drop out.

SMPTE Time Code Jump This error means that a discontinuity occurred in the time code. For example 00:01:00:01, 00:01:00:02, 00:03:00:12, 00:03:00:13 would cause this error. Jumps typically occur when you fast forward or rewind the time code source.

If you have multiple AD516s or AD1012s, use Preferences to select the card to be the source of time code monitored.

Procedures:

SMPTE Capture

SMPTE Capture allows you to determine the exact frame number of a specific event. By clicking the space bar when the SMPTE Monitor window is active, you can stop and start the SMPTE monitor.

1. Load SMPTE Generator and SMPTE Monitor from Applications Menu.

2. Start the SMPTE Generator by clicking **Play.**

3. IMPORTANT - Click on the SMPTE monitor window to make it active.

4. Select Freeze Display from the SMPTE Monitor Menu to capture the SMPTE Monitor display. Note that the keyboard shortcut for Freeze Display is *A*-F

5. Hit *A*-F again or select the freeze menu option to resume monitoring.

You can also capture specific time frame in the Cue List by using the TC Add Option.

Name: **Studio16**

Class: Shell Command

Format: **`studio16`**

Description: The Studio16 command is used to launch the Studio 16 environment. After Studio 16 loads itself, it will display an About window containing the program revision number and other specifications about your setup.

The Studio 16 **Screen** command will allow you to load Studio 16 into a WorkBench screen or another screen.

Procedures:

Opening Studio 16 on a WorkBench Screen

Loading Studio 16 to a WorkBench screen allows you to take advantage of monitors with higher resolution that only work with WorkBench.

From Shell, type **`studio16 Screen`** ***`WorkBench`***

You should change your WorkBench screen to 8 colors using WorkBench preferences.

Opening Studio 16 on any Screen

Open Studio16 on any screen by using the above command, but replace WorkBench with the first few letter of the alternate screen's title. The title is located in the upper left corner on the screen's title bar.

For example, if you have an application who opens a screen with the text "ABC Software Corp. (c) 1992" in the title bar, you could cause Studio16 to open onto this screen by typing:

```
studio16 Screen ABC
```

Studio 16 prefers an 8 color screen, but it will try to open on anything. Also, the color mapping of the screen's application may not match Studio 16's.

Name: **StudioClose**

Class: Shell Command

Format: **`StudioClose`**

Description: StudioClose is a Shell command that removes all Studio 16 modules from memory. It's just like selecting Quit. See StudioPlay for an illustration.

Name: **StudioOpen**

Class: Shell Command

Format: **`StudioOpen`** *`ModuleName`*

ModuleName represents the name of a Studio 16 module. (Optional Parameter)

Description: StudioOpen is a Shell command that will open the Studio16 library and load device drivers and utilities plus an optional Studio 16 module into memory. See StudioPlay for an illustration.

Name: **StudioPlay**

Class: Shell Command

Format: `StudioPlay` *`SampleName`*

SampleName represents the name of a Studio 16 sample. (It should be a complete path, i.e. Audio:DoorBell)

Description: A Shell command that allows you to play sounds from the AD1012 without loading Studio 16. It may also be used to incorporate 16 bit samples in your own programs.

Procedures:

Playing Multiple Sounds

If you plan to do multiple StudioPlay commands, and if you want to maximize the speed at which they respond, you can use the following sequence:

```
StudioOpen
StudioPlay <SampleName>
StudioPlay <SampleName>
StudioClose
```

The command **StudioOpen** will load Studio16 device drivers to RAM. **StudioClose** removes them.

Name: **StudioQuery**

Class: Shell Command

Format: `StudioQuery`

Description: StudioQuery will display the current SMPTE time code in the format HH:MM:SS:FF. For a quicker response, precede the command with StudioOpen, and follow it with StudioClose. See StudioPlay for a similar example.

Name: **StudioStop**

Class: Shell Command

Format: **`StudioStop`**

Description: StudioStop is a Shell command that can used to stop currently playing sounds from the Shell. See StudioPlay for an illustration of Shell Commands.

Name: **StudioWait**

Class: Shell Command

Format: **`StudioWait`** *`HH:MM:SS:FF`*

Description: StudioWait will wait for a specified SMPTE time to occur. This is useful for synchronizing SHELL or AREXX scripts, AmigaVision, CanDo, and other programs with SMPTE time code. To use StudioWait in a program, that program must be capable of executing a DOS or SHELL command. StudioWait can be interrupted with a ^C or the 'break' AmigaDOS shell command. StudioWait loads the device drivers when it is first executed. For maximum response, you can use the sequence:

```
StudioOpen
StudioWait <time code>
StudioWait <time code>
StudioClose
```

Name: **Time**

Class: Application Module

Description: The Time module displays the current system time of your Amiga. Figure 1. This module is resizeable which makes it easy to place on the Studio16 screen. Refer to you Amiga manual for instructions on setting the system time.

Figure 1. Time

Name: **Tiny Mixer**

Class: Application Module

Description: This module controls the volume of each of the playback channels as well as the input and output channel. The volume may be changed before, during, or after sample playback. The input volume slider is used to control the volume of the monitor channel. That is, the amount of sound passing from the input of the card to the output of the card. Use a mixer when you need to adjust a sample's volume in realtime.

The Tiny Mixer features digital volume controls with a range from +6dB to -40dB in 1dB increments and -00dB (-∞ or, no volume). Figure 1.

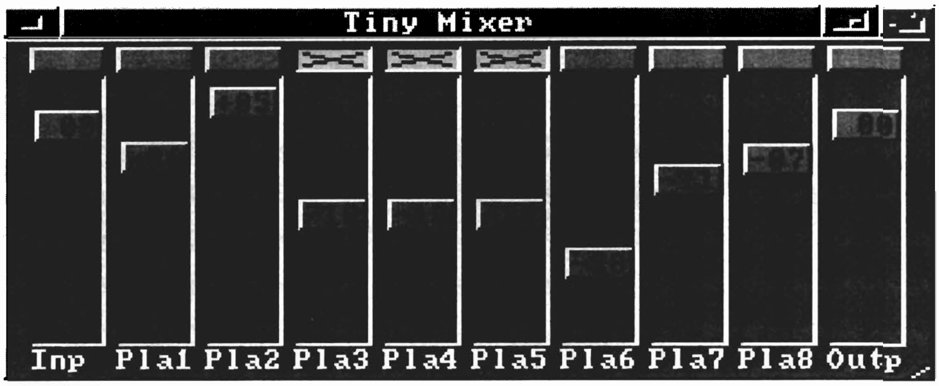


Figure 1. Tiny Mixer

The Tiny Mixer has three advantages over the main Mixer:

- Tiny Mixer has a multiple channel select feature. Click on the active buttons at the top of the channel sliders to select multiple channels that can be adjusted simultaneously.
- Tiny Mixer takes up less memory and CPU processing time than the standard Mixer module. It should be use on smaller or slower computer systems.
- Tiny Mixer is resizeable and will fit just about anywhere on a crowded Studio16 screen.

AD516 ONLY - The Tiny Mixer does not have a pan adjustment.

Menu: **Tiny Mixer**

The only menu option for Tiny Mixer contains the channel selector. Use your mouse to select which channels to show in the mixer.

Name: **Transport**

Class: Application Module

Description: The Transport module can be used as a digital multi-track recorder. It operates in one of three modes:

- Playback sample aligned sounds
- Playback sample aligned sounds while recording the input of the card
- Playback sample aligned sounds while recording the output of the card

You can record as many tracks as you want and then digitally ping-pong them into submixes. Then you can combine your submixes into a complete song. You can even use the transport to record your Cue Lists into a single file.

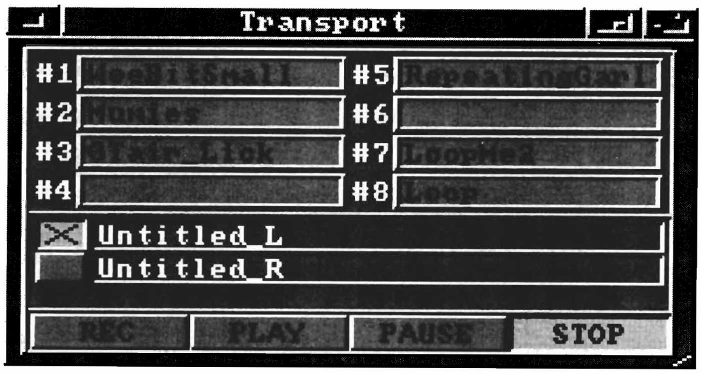


Figure 1. AD516 Transport

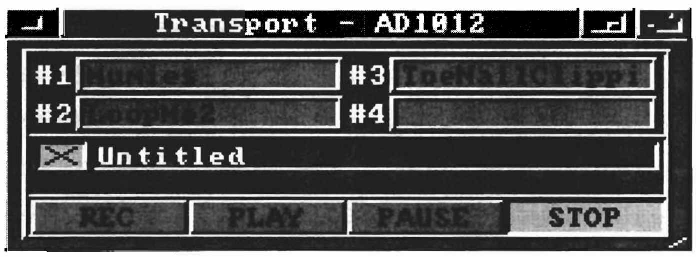


Figure 2. AD1012 Transport

Layout: Transport varies depending on whether you have an AD516 or AD1012 installed. With the AD516 you will have 8 playback tracks and 2 record tracks,. With the AD1012 installed, transport has 4 playback tracks and 1 record track.

The playback tracks are at the top of the window indicated by numbers. #1-#8 or #1-#4, depending the AD516 or AD1012. These tracks contain the samples that will be played back.

AD1012 ONLY - When the record track is enabled, the number of playback tracks is reduced to 3.

Below are the record track(s). Before recording, a record track must be activated. Click the active box next to its name to activate it.

Along the bottom of the window are your record statistics and controls. Use the controls to start the playback and the recording of samples. The **Pause** button pauses recording and playback. In order to record, both the **Record** and **Play** buttons must be selected.

Procedures:

Playing Two Sounds at Once

1. Select two samples in the Open List with the same sampling rate.

2. Load Transport from the Applications Menu, ^ T.

3. Click and drag the samples from Open List to the playback tracks of Transport.

4. Click the **Play** button.

Record a Sample

1. Clear all the Playback tracks by clicking Clear from the Transport Menu.

2. Connect your audio source, and load Mixer. To monitor the incoming audio, increase the input level to +0dBs.

3. Activate a record track and type in a name for the new sample.

AD516 ONLY - To record a stereo sample, activate both the record tracks by clicking their active boxes.

4. Set the Transport Menu Options as follows:

   ✓ Record Input
   ✓ Manual Start
   ✓ Manual Stop
   ✓ Auto Volume
   ✓ Auto Rate

5. Select the **Play** and **Record** button to begin recording.

6. The status line will indicate the size of the sample as it is being recorded, and the amount of remaining free disk space.

7. Click **Stop** to stop recording.

8. Check Open List for the new sample.

Recording and Playing Samples at the Same Time

1. Drag a sample from the Open List to a playback track.

2. Activate a record track and type in a name for the new sample.

AD516 ONLY - To record a stereo sample, activate both the record tracks by clicking their active boxes.

3. Set the Transport Menu Options as follows:

   ✓ Record Input*
   ✓ Manual Start
   ✓ Auto Stop
   ✓ Auto Volume
   ✓ Auto Rate

   * Record Input records just the input, it does not mix in the playback sample . Record Output, would record both the playback and the input audio.

4. Select the **Play** and **Record** button to begin recording.

5. The status line will indicate the size of the sample as it is being recorded, and the amount of remaining free disk space.

6. Recording will stop automatically, as soon as the playback sample has finished.

7. Check the Open List for the new sample.

Mixing Two Sounds Together (Ping-Ponging or Bouncing Tracks)

1. Select two samples in the Open List with the same sampling rate.

2. Click and drag the samples from Open List to the playback track(s) in Transport.

3. Activate a record track and type in a name for the new sample.

AD516 ONLY - To record a stereo sample, activate both the record tracks by clicking their active boxes.

4. Set the Transport Menu Options as follows:
   ✓ Record Output
   ✓ Manual Start
   ✓ Auto Stop
   ✓ Auto Volume
   ✓ Auto Rate

5. Load Mixer and make sure the Input level is set all the way down to -00dB, close the Mixer.

6. Select the **Play** and **Record** button to begin recording.

7. The status line will indicate the size of the sample as it is being recorded, and the amount of remaining free disk space.

8. Recording will stop automatically, as soon as both samples have finished playing.

9. Check the Open List for the new sample.

If Flashing Screens Occur During Mixing
(For slower systems with the AD1012)

The following technique will allow you to mix three samples at a high sampling rate without overloading your system. Often you are mixing samples together because your system can't handle their simultaneous realtime playback. The following steps show how to mix three samples together into one new sample, even if you can't play the three samples simultaneously. The trick is mix at a low rate and then raise the rate of the new sample after the mix is complete.

1. Drag a few samples from Open List into the Transport Playback tracks. Note that all samples require the same default sampling rate. Check the Set Sample Parms Option in the editor for a sample's default rate .

2. From the Transport Menu, select Manual Rate. In the Options - Handler menu, select a rate lower that the default rate. For example, if the samples were originally recorded at 44,247KHz, change the rate to 22,026KHz. You can use any rate, even rates lower than 22,026KHz. If you click **Play** now, the samples will play at a very slow rate.

   The samples have not been altered though, and will play normally from other modules.

   After lowering the playback rate you will be able to change it back to the original rate without any adverse effects.

3. Now, mix the samples using the previous Mixing technique (Steps 1-10), but set the Transport Menu Options as follows:
   ✓ Record Output
   ✓ Manual Start
   ✓ Auto Stop
   ✓ Auto Volume
   ✓ Manual Rate

   During the mix, the pitch will sound too low, but the mix is occurring correctly.

4. After the samples have been mixed, open the new sample in an Editor and select Set Sample Parms from the Option Menu. Change the Rate to original sampling rate of the individual samples (44,247KHz in the above example).

5. Your new sample will now sound as if you mixed your samples at normal speed.

Increasing the Level of a Submix

If, after a mix, you determine that a channel's level was too low, you can mix that one channel in a second time to increase its level.

From the diagram below (figure 3), notice that Track 1, 2, and 3 were mixed together using the previous ping-pong technique. The sample was called SubMix#1. The level of Track 3 was too low. Rather than remix the samples, you can just boost the level of track 3 by remixing it in again.

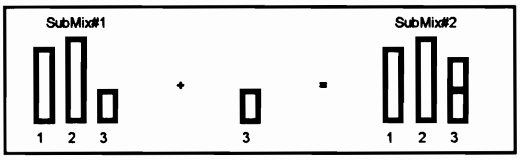


Figure 3. Increasing the Level of a Submix.

1. In the Transport Option Menu, select Clear to remove any existing playback entries.
2. From Open List, drag Track 3 and SubMix#1 into Transport Playback Tracks.
3. Type SubMix#2 in a record track, and make it active.
4. Set the Record Parameters as follows:
   ✓ Record Output
   ✓ Manual Start
   ✓ Auto Stop
   ✓ Auto Volume
   ✓ Auto Rate
5. Load Mixer and make sure the Input level is set all the way down to -00dB, close the Mixer.
6. Select the **Record** and **Play** to begin recording.
7. The status line will indicate the size of the sample as it is being recorded, and the amount of remaining free disk space.
8. Recording will stop automatically, as soon as both samples have finished playing.
9. Play the new mix from Transport or Open List.

Recording the Cue List Output

1. Load the Cue List and SMPTE Generator module from the Applications Menu.
2. Enter or Load a Cue List, refer to the Cue List Reference Section for details.
3. Test the Cue List by clicking Play on the SMPTE Generator. (Make sure the Cue List is **ON**.)
4. After samples play, reset the SMPTE Generator by clicking **Return** (set the return about 1 second before the first sample triggers).

5. In Transport, set the Record Parameters as follows:
   - ✓ Record Output
   - ✓ SMPTE Start
   - ✓ SMPTE Stop
   - ✓ Auto Volume
   - ✓ Auto Rate

6. Select the SMPTE Start and Stop from SMPTE in Transport Options Menu. Type in a start and stop time code. The start time should be 1 second before the first sample plays, and the end time should be about 1 second after the last sample stops playing. Inspect the end times to find the sample that will finish last.

7. Load Mixer and make sure the Input level is set all the way down to -00dB, close the Mixer.

8. Before you can record, you must click Record and then the Play button in Transport. This will bring up the Transport SMPTE window, Figure 4. The Start and End times are entered from the Transport's Options Menu. You only need to click Play if you want to ignore the SMPTE trigger and begin playing immediately. Otherwise, Transport will trigger when the SMPTE time is received.

Figure 4. Transport SMPTE

9. Click the **PLAY** button on the SMPTE Generator to trigger the Cue List, the recording will begin when the start time is reached.

10. The status line will indicate the size of the sample as it is being recorded, and the amount of remaining free disk space.

11. The recording will stop when the end time is reached, or if stop is selected.

Gadget:

Playback Tracks

The number of playback tracks depends on whether you have an AD516 or and AD1012 installed. The AD516 will provide 8 playback tracks, the AD1012 has 4 playback tracks.

Samples or Regions are added to the playback track by dragging them from the Open List. For a region list in Open List, select Show Regions from the Open List menu. You can also drag in regions to play from the Editor Regions List. Select Show Regions in the Editor's Option menu.

AD1012 ONLY -	When recording with the AD1012, playback track #4, will be disabled, because the recording will actually take place on this fourth track.

Record Tracks

This field contains the base file name for any sample you record. If the sample name is already being used by another sample, the transport will append a number to the end of the name. The default is "Untitled".

Record tracks must be activated before they can be used for recording. Click to the left of the record track to activate it.

To change the name of a record track, simply type in a new one.

Size: When recording, Size: will update with the current size of the sample as it records.

Free: Lists the free space available in the record path. This number will update with the remaining free space during recording. To change the record path, load Open List, and make another directory active.

Record Click this button to begin a recording (then click the play button immediately afterwards).

Play Cause the samples in the playback tracks to play. Also, it must be selected in addition to the record button, for transport to record.

Pause Will pause any playing and/or recording.

Stop Will stop the playing and/or recording. Will also stop samples playing from other modules.

Menu: **Transport**

Record Input Records the input of the card exclusively. You may playback samples while you record but they will not effect the recording.

Record Output Records the output of the mixer (whatever you hear coming out of the card). This allows you to ping-pong or bounce multiple playback tracks into one track by combining the several samples together into submixes and then playback the submixes. See Figure 3, the Mixer Diagram for an illustration of recording from the Input and Output Tap.

Manual Start Begins the recording whenever the record and play buttons are hit.

SMPTE Start Causes the recording to begin when a specific time frame is reached. The time frame is set in the SMPTE Menu Option.

Manual Stop Allows you to stop recording by clicking the Transport's Stop button.

SMPTE Stop Causes the recording to stop when a specific time frame is reached. The time frame is set in the SMPTE Menu Option. You may override auto stop by manually clicking Transport's **Stop** button.

Auto Stop Automatically stops recording after the longest playing sample ends.

Manual Volume Plays at the Volume level setting of the Mixer.
Auto Volume Plays at the default volume level of the sample. (Set in the Editor - Set Sample Parms, usually +00dB).

Manual Rate Plays at the Sampling Rate as set in Handler Parameters.
Auto Rate Plays at the default sampling rate of the samples. (Set in the Editor - Set Sample Parms).

See the Recorder Reference Section for more details on setting a sampling rate.

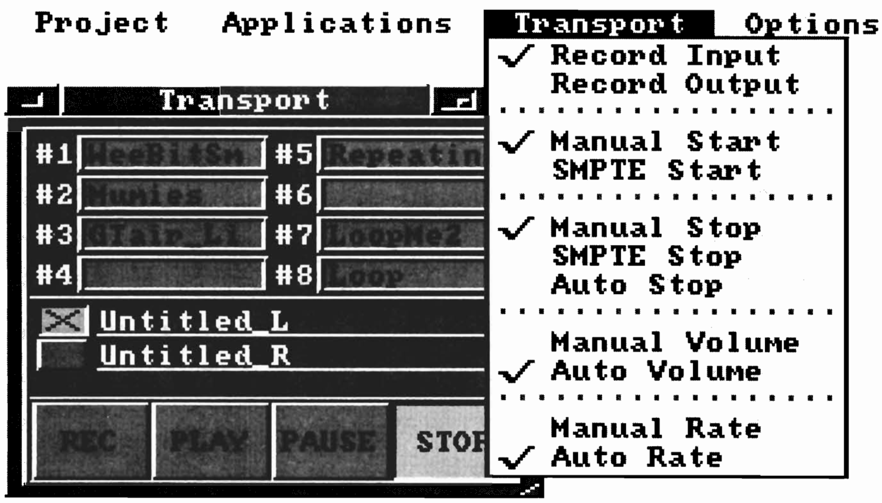


Figure 5. AD516 Transport and Transport Menu

Menu: **Options**

Clear Removes all samples from the playback tracks.

Handler Allows you to set a manual rate that's activated when Manual Rate is selected in the Transport Menu. See the Reference Section on Recorder for explanation of sampling and filtering rates.

If you have trouble selecting a sampling rate that is available in the Recorder, use the arrow buttons or click in the slider to the left or right of the knob. This will display all available increments, just dragging the knob does not show them all.

SMPTE: **Start** Sets the start time of a recording. It is only accessed if the Transport Menu is set to SMPTE Start. The start time should be about 1 second before the first sample plays to allow for preloading.

Stop Sets the stop time of a recording. It is only accessed if the Transport Menu is set to SMPTE Stop. The end time should be set to about 1 second after the last sample stops playing if you are recording a Cue List. Inspect the end times to find the sample that will finish last and add 1 second.

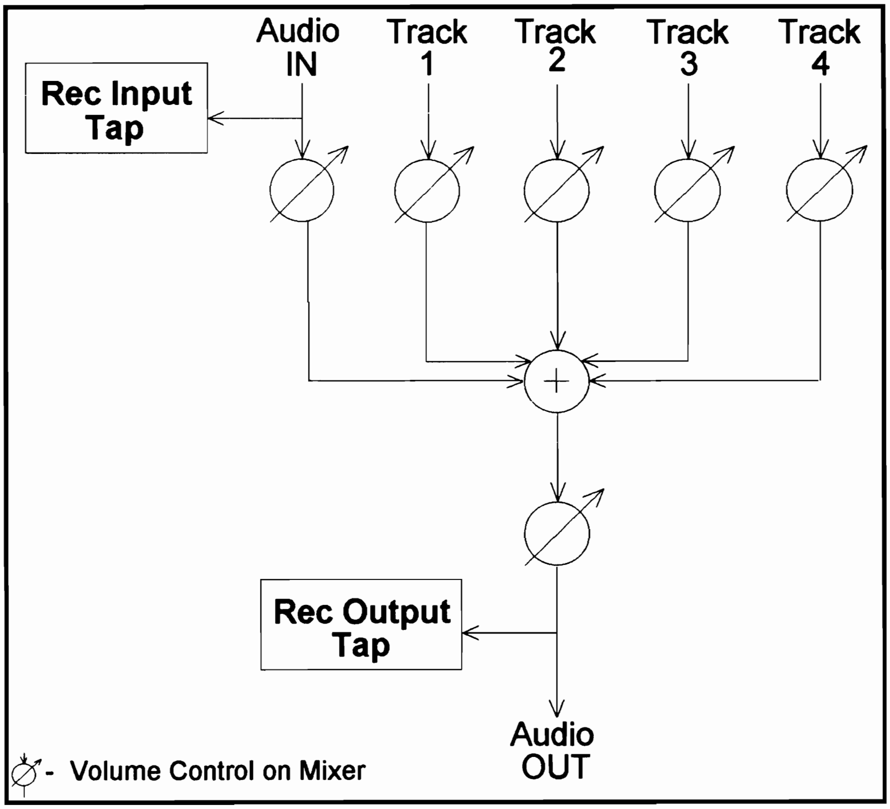


Figure 3. Mixer Schematic

Name: **Utility**

Class: Utility Module

Description: Utility is responsible for loading and saving files and is used when playing sounds. It does not require user access.

Appendix A

Technical Support

If you have any problems with your Studio 16 system, please note the following instructions.

1. If you have a read/write error on your installation floppy disk, call SunRize for replacement.
2. Check the program disk for a **Read_Me** file. This file contains any recent developments and update information that is not included in this manual.
3. Read Chapter 7 - Troubleshooting.
4. If you suspect a malfunctioning AD516 or AD1012, call SunRize for instructions.

Note: When calling SunRize for technical support, please have the Studio 16 software version number handy. It is listed in the About window.

SunRize Industries Technical Support can be reached at (408) 374-4962. Technical Support is available Monday through Friday from 9:00 a.m. to 5:00 p.m. PST.

SunRize Industries
2959 S. Winchester Blvd., Suite 204
Campbell, CA 95008 USA
Tel: (408)-374-4962
Fax: (408)-374-4963

European users may contact their local distributor.

Finland
Power Computers
Malininkatu 2............. Tel: 018 781 8992
SF-15100 Lahti ....Fax: 018 781 8993

France
Storm Media Production
6 Bis Rue de Candie................ Tel: 01 43 57 46 57
75011 Paris ....Fax: 01 48 05 75 53

Germany
Advance Systems
Homburger Landstraße 412.... Tel: 069 5 48 81 30
6000 Frankfurt 50..................Fax: 069 5 48 18 45

Italy
Applied Peripherals & Software
Via Giovanni XXIII, 37............ Tel: 0432 759 264
33040 Corno di Rosazzo, Udine Fax: 0432 759 264

Spain
PiXeLSOFT
Felipe II, 3 Bis........................... Tel: 088 71 27 00
34004 Palencia...........................Fax: 088 71 27 00

Sweden
Display Data
Äskan 1087................................. Tel: 0457 503 80
S-370 11 Backaryd......................Fax: 0457 503 84

Switzerland
Microtron
Bahnhofstrasse 2....................... Tel: 032 87 24 29
2542 Pieterlen............................Fax: 032 87 24 82

United Kingdom
HB Marketing
Unit 3, Poyle 14, Newlands Dr. Tel: 0753 686000
Colnbrook, Berks SL3 0DX ....Fax: 0753 680343

Appendix B

Hardware Specifications

AD516

- 5 tracks with 68000 at 44KHz
- 8 tracks with 68030 at 44KHz
- Simultaneous Record and Playback
- Frequency Response 15Hz - 22KHz (-3dB)
- ADSP 2105 sound coprocessor (rated at 10MIPS, 100ns instruction execution time)
- LTC SMPTE time code reader (24,25,29.97,30 fps, drop and non-drop)
- Dual 16 bit delta-sigma A/D converters with digital anti-aliasing filters
- Dual 16 bit delta-sigma D/A converters with digital anti-aliasing filters
- 64 times oversampling
- 14 different sampling rates from 5.5KHz to 48KHz
- > 85dB Dynamic Range
- THD+N < .0095%
- Stereo RCA jacks for unbalanced line level (2 Volt RMS) audio inputs and outputs
- Input resistance 50K
- Output resistance < 1 ohm
- Input digital gain levels: 16
- 256K of fast static RAM
- 8K of ROM
- Zorro II AutoConfig card
- Dimensions: 13.3 X 4.2 inches
- Power supplied by Amiga
- List Price $1495

AD1012

- 1 tracks with 68000 at 44KHz
- 4 tracks with 68030 at 44KHz
- Simultaneous Record and Playback
- Frequency response 20Hz - 20KHz (-3dB)
- ADSP 2105 sound coprocessor (rated at 10 MIPS, 100ns instruction execution time)
- LTC SMPTE time code reader (24,25,29.97,30 fps, drop and non-drop)
- 12 bit linear analog-to-digital converter with sample and hold
- 12 bit linear digital-to-analog converter with FIFO buffer
- Two eighth order anti-aliasing filters
- Adjustable sampling rate up to 100KHz
- >70dB Dynamic Range
- THD+N < 0.04%
- Mono RCA jacks for unbalanced line level audio input and output
- Input resistance 50K
- Output resistance < 1 ohm
- Input digital gain levels: 100
- 64K of fast static RAM, expandable to 256K
- 8K of ROM
- ZORRO II AutoConfig card
- Dimensions: 13.3 x 4.2 x 0.6 inches
- Power supplied by Amiga
- List Price $595

Appendix C

Bars & Pipes Professional

Your SunRize Studio 16 software integrates seamlessly into Bars&Pipes Professional, a popular MIDI composition and sequencing program. Bars&Pipes is not included with Studio 16--you must purchase it separately. Contact your Amiga dealer or Blue Ribbon Sound Works for more information. Blue Ribbon has written four modules that load directly into Bars&Pipes Professional and turn your Amiga into a complete music and audio production environment. These modules are included free with Studio 16.

Once you've installed these modules (Tools and Accessories in B&P Pro parlance) into Bars&Pipes Professional, you can:

- Synchronize your MIDI composition with SMPTE via the SMPTE input in the SunRize card.
- Access all Studio16 features and windows within the Bars&Pipes Professional screen.
- Cue samples to MIDI notes. This is extremely useful for cueing sound effects to music as well as video.
- Record and play back a virtual digital track in sync with your MIDI tracks.

To install the SunRize Tools in Bars&Pipes Professional, copy the tools from the "Bars&Pipes/Tools" drawer on your Studio 16 Master disk #2 into your Bars&Pipes Tools drawer and copy the accessories from the "Bars&Pipes/Accessories" drawer on your Studio 16 Master disk #2 into your Bars&Pipes Accessories drawer.

From within Bars&Pipes Pro, open the Tool window and select the menu option "Load Tool..." ("Install Tool..." in B&P Pro 2.0) to load in the desired tools. To load the accessories, open the Accessories window and select "Load..." for each accessory you wish to load.

For technical support regarding Bars & Pipes Professional, please contact Blue Ribbon SoundWorks - 1605 Chantilly Dr., Suite 200, Atlanta, GA 30324 USA, Tel: (404) 315-0212

ACCESSORIES:

NAME: SunMPTE

DESCRIPTION: The SunMPTE Accessory allows Bars&Pipes Pro to synchronize to SMPTE input entering the SMPTE port on AD516 or AD1012, or generated by the SunRize SMPTE Generator module.

When you double-click on this accessory's icon, a control window opens. At the top the current SMPTE time is displayed. Underneath are two buttons, Activate and Run Lock. If the Activate button is depressed Bars&Pipes Pro will receive SMPTE and will start and stop in sync with SMPTE input. If both the Activate and Run Lock buttons are depressed (default) Bars&Pipes Pro continuously updates its internal time based on SMPTE input events, ensuring proper synchronization.

NAME: SunSet

DESCRIPTION: The SunSet Accessory allows you to run all the SunRize Studio 16 modules within the Bars&Pipes Pro screen. When this icon is double-clicked the Studio 16 Instance window will open, allowing you to access all of the Studio 16 software.

The SunSet Accessory also allows you to have a unique environment setup within Bars&Pipes Pro. When the Instance window opens, there will always be a module named "B&PSavePrefs". Selecting this module saves all preferences in the Master Preferences (except the color map) and the size, position, and configuration of all modules currently open. This information is saved independently of the normal Studio16 preferences. The files used to save this information are in the S: directory, Studio16BPP.config holds the defaults from the Master Preferences, and BPPInstance holds the configuration of all the open modules.

Bars&Pipes Pro keeps track of whether the Instance window is open and if Quit is selected to leave Bars&Pipes Pro and the Instance window is open, all other modules will be automatically closed. If you close the Instance window, make sure that you close all other Studio16 windows before leaving Bars&Pipes Pro.

NOTE: Using this accessory may have unpredictable results if you are already running Studio16 on another screen.

TOOLS:

NAME: SunRize Out

DESCRIPTION: The SunRize Out tool translates MIDI note events into samples to play out of the SunRize audio card. You may assign a different sample to each of the 128 MIDI notes. You have the option either of playing each sample out to its full duration or stopping playback when a Note Off is received.

SPECIAL TYPE: Output

USAGE: PipeLine

CONTROLS: The Sample button states the name of the sample associated with the note displayed in the Note slider, or "Undefined" if no sample is associated with that note. The Note slider allows you to select the MIDI note to be displayed.

To the right of the sample button is the Load button. Clicking on this button brings up a file requester so that you may select a sample to assign to the current note, as displayed in the Note slider. Sample file names are stored internally with their full path name, so if a sample is renamed or moved from one directory to another it will invalidate that entry in the sample list.

To the right of the Load button is the Record button. Clicking on this button loads the Studio16 Recorder module. This button is provided as a convenient way to record new samples quickly without having to load all of the Studio16 software.

To the right of the Note slider under the Record button is a button labeled Test. Clicking on this button plays the current sample, based on the current note.

At the bottom right of the window is a button labeled Force Duration. If this button is depressed, samples are clipped to be equal to the note length only. Otherwise each sample is played out in its entirety.

The Driver button at the bottom of the window displays the current SunRize device drive that is used to play back samples. If you have more than one SunRize card, clicking on this button will bring up a scrolling list of all available drivers to choose from.

Once this tool is set up, whenever it receives a note from the pipeline that corresponds to a sample, it will begin playing. In all cases, pressing Stop from the Transport Controls will stop the sample playing.

The SunRize Out Tool is sensitive to both velocity and Control Change 7 (Volume) for determining the volume of the sample. When used with the AD516 board, Control Change 10 (Panning) is used to determine the panning of the sample.

NAME: SunRize Virtual Track

DESCRIPTION: The SunRize Virtual Track tool acts as a separate track in the Bars&Pipes Pro environment. It allows you to record and playback digital audio via your SunRize card, synchronized to begin at any specified point in time.

SPECIAL TYPE: None

USAGE: PipeLine

CONTROLS: Two different setups may appear, depending on whether the current handler is the AD1012 handler of the AD516 handler. If you have both, the window will change its setup when you select a new handler.

AD1012: The string defined by Name: is the full path name to the sample that is to be recorded or played back. The default is "<the Studio16 default path>Untitled". The Load button to the left opens up a file requester so that you may find a previously recorded sample; the name of the file selected is put into the Name: string. While recording, if a sample of the same name already exists, that sample is deleted.

Directly under the "Load" button is the "Monitor" button. This button merely turns on and off monitoring of the input stream.

Several sliders and the Auto Filter button control the parameters for recording samples. The Gain slider adjusts the input gain, and the Rate and Filter sliders determine the sampling rate and filter setting. With Auto Filter on (the default), the Filter automatically adjusts to a good value based on the rate. None of these options has any effect on playback of samples.

Below the filter setting is the Start time. This is the time that the sample will actually start playing or recording. By default this time is in music time, but by selecting the SMPTE button to the left of the start time, the time may be specified in SMPTE format. Because of disk access time it is not always possible to start a large number of samples playing or recording at the same time, so staggering the start times is recommended when playing more that two samples at a time.

The Meters button in the bottom left of the screen is a convenience provided to open up the Studio16 Meters window within the Bars&Pipes Pro screen. This will help you to adjust the input levels to get maximum response when recording samples.

The Driver button at the bottom of the window displays the current SunRize device drive that is used to play back samples. If you have more than one SunRize card, clicking on this button will bring up a scrolling list of all available drivers to choose from.

To playback a sample using the SunRize Virtual Track Tool, simply place this tool in a pipeline, type or use the file requester to assign the name of the sample, optionally modify the start time to begin playing the sample, and then press start from the Bars&Pipes Transport Control. The sample will begin playing at the requested time (or as close as possible if disk access time is slow) and will stop when either the sample ends or Stop is pressed in the Transport Controls. If the song begins playing after the requested start time, the sample will begin at the appropriate point starting at the next measure boundary. If the track that the tool is in is muted, the sample will not play.

To record a sample using the SunRize Virtual Track tool, set up the name and start time as for a playback sample, but also set up the desired rate, filter, and gain settings. Use of the Monitor and Meters Window may come in handy to determine the proper settings. Just as in a regular MIDI track for Bars&Pipes Pro, the track that contains the SunRize Virtual Track Tool must be in Record mode and the Record button in the Transport Controls must be set for recording to take place. Once the record mode is set, press Start in the Transport controls and recording will begin at the start time specified in the tool. If the song begins after the requested start time, the sample will not be recorded. The only way to stop recording is to press Stop in the Transport Controls.

The SunRize Virtual Track Tool is sensitive to Control Change 7 (Volume), and if placed in an output pipeline can be used with MixMaestro.

AD516: The two strings defined by "Left:" and "Right:" are the full path names to the samples that are to be recorded or played back. The defaults are "<the Studio16 default path>Untitled_L" and "<the Studio16 default path>Untitled_R". The "Load" button to the left of each string opens up a file requester so that you may find a previously recorded sample; the name of the file selected is put into the string immediately to the right of the "Load" button selected. While recording, if a sample of the same name already exists, that sample is deleted.

Directly under the "Load" buttons are two buttons labeled "L" and "R". These buttons define which samples of the two listed (Left or Right) will be recorded to or played back from. Either one of these buttons or both may be selected. If only one sample is being recorded, it is recorded as a mono track (default pan is centered). If both are selected the left sample is recorded with pan set full left and the right sample is recorded with pan set full right.

Directly under the "L" and "R" buttons is the "Monitor" button. This button merely turns on and off monitoring of the input stream.

The "Gain:" and "Rate:" sliders control the parameters for recording samples. The Gain slider adjusts the input gain, and the Rate slider determines the sampling rate. None of these options has any effect on playback of samples.

Below the "Rate:" slider is the Start time. This is the time that the sample will actually start playing or recording. By default this time is in music time, but by selecting the SMPTE button to the left of the start time, the time may be specified in SMPTE format. Because of disk access time it is not always possible to start a large number of samples playing or recording at the same time, so staggering the start times is recommended when playing multiple samples at one time.

The Meters button in the bottom left of the screen is a convenience provided to open up the Studio16 Meters window within the Bars&Pipes Pro screen. This will help you to adjust the input levels to get maximum response when recording samples.

The Driver button at the bottom of the window displays the current SunRize device drive that is used to play back samples. If you have more than one SunRize card, clicking on this button will bring up a scrolling list of all available drivers to choose from.

To playback a sample using the SunRize Virtual Track Tool, simply place this tool in a pipeline, type or use the file requester to assign the name of the sample, optionally modify the start time to begin playing the sample, and then press start from the Bars&Pipes Transport Control. The sample will begin playing at the requested time (or as close as possible if disk access time is slow) and will stop when either the sample ends or Stop is pressed in the Transport Controls. If the song begins playing after the requested start time, the sample will begin at the appropriate point starting at the next measure boundary. If the track that the tool is in is muted, the sample will not play.

To record a sample using the SunRize Virtual Track tool, set up the name and start time as for a playback sample, but also set up the desired rate and gain settings. Use of the Monitor and Meters Window may come in handy to determine the proper settings. Just as in a regular MIDI track for Bars&Pipes Pro, the track that contains the SunRize Virtual Track Tool must be in Record mode and the Record button in the Transport Controls must be set for recording to take place. Once the record mode is set, press Start in the Transport controls and recording will begin at the start time specified in the tool. If the song begins after the requested start time, the sample will not be recorded. The only way to stop recording is to press Stop in the Transport Controls.

The SunRize Virtual Track Tool is sensitive to Control Change 7 (Volume) to determine the volume of both samples. Panning via Control Change 10 is also supported. For samples recorded Mono panning works normally, but for stereo samples (two samples playing simultaneously), panning reacts as if the Right sample is supposed to be full right and the Left sample is supposed to be full left. Panning to the right will move the Left sample to the right; panning to the left will move the Right sample to the left. If placed in an output pipeline MixMaestro can be used to control both volume and panning.

NOTE: Disk access time is a limiting factor in playing back and recording samples. Many samples playing at once and/or slow hard drives can cause samples to get out of sync with Bars&Pipes Pro. In many cases you may be able to get around these problems by optimizing our buffer sizes in the Master Preferences or moving critical samples to RAM:. In other cases the only solution may be to stagger your samples so that fewer are playing at the same time.

Appendix D

Introduction to Audio Post Production

A BEGINNERS GUIDE TO USING STUDIO 16 IN AUDIO POST PRODUCTION FOR FILM AND VIDEO

Usually when people make movies, they put such an emphasis on the shooting that they consider the sound a mere technical thing, but sound and music are the great seducers in film.

Francis Ford Coppola

In September 1991 I was invited by Commodore Computers, Norway to visit the annual Amiga-users Symposium in Oslo (ASO). One of the main events was the introduction of a professional hard-disk recording system for the Amiga 2000 / 3000 series, soon to released. We were introduced to Studio 16 from SunRize Industries. Surely a computer program that recorded and played back 12 bit digital audio was not a sensation in 1991, but where most programs were oriented towards music production, this program seemed tailored for audio production. I called SunRize Industries and within two weeks I could plug my AD516/AD1012 card into my Amiga 3000, install software version beta 0.52, and start exploring this fantastic new way of sound editing.

Within two hours I was in love, and I still am. And I think you will be too!

Oslo, Norway July 17.1992

Øistein Boassen.

Introduction to Audio Post Production

Table of Contents

1. BEGINNERS GUIDE TO AUDIO POST PRODUCTION

1.1 INTRODUCTION

For years and years there were actors like Charlie Chaplin and Buster Keaton who could make the whole world laugh and cry - without using a sound, only pictures. Today you can hardly find a movie theater without a Dolby Stereo sound system, and all major TV Stations are transmitting stereophonic sound. Sound technicians have always been dissatisfied with the traditional cut and splice editing method. First, tape editing it is not very accurate and second it does not tolerate many mistakes before the tape is damaged. The frustrations of having to waste valuable post production time on sound transferring instead of creative editing have often taken some of the joy away from sound editing.

In resent years, since Dolby Stereo was introduced, sound has gradually become more and more important as the audience has become more and more aware of it. Today's professional film and video producers are very concerned with sound, but as their demand for good sound is gradually getting stronger, their audio post-production budgets are gradually getting smaller! Professional audio production and editing are traditionally very time-consuming, thus quite expensive. Digital sound editing is incredibly speedy and accurate; therefore, less expensive than traditional analog sound editing. These are just some of the reasons why digital sound processing is rapidly becoming the most popular sound editing tool in history.

In this chapter we look at basic audio for video and see how digital sound recording and editing can replace traditional methods such as tape recorders and editing-tables.

1.2 A QUICK TOUR OF TRADITIONAL AUDIO POST PRODUCTION

Film 16 / 35 mm.

In motion pictures there has never been any real trouble in synchronizing sound and picture although the picture and the sound have always been recorded separately. In the beginning of each scene there is a clapper recorded both on the film and on the audio tape. When the shooting is over, the film is sent to the developing laboratory, and the sound is transferred to a specially perforated audio tape. The sprocket-holes used for film transport are also used on the magnetic audio tape. To synchronize the sound with the picture one uses an editing table that shows the picture and two channels of audio at the same time. The audio is synchronized by looking at the picture and identifying the frame where the clapper goes bang, then listening to the audio tape until the sound of the clapper going bang is heard, and marking that place with a marker. Once the audio is locked in position with the film, the sound is synchronized.

In 16 mm film there is only one perforation per frame, making the synchronicity limited to one frame accuracy . 1/25 (1/30) of a second. 35 mm film offer four points of synchronization for each picture frame. That is about 1/100 of a second accuracy. With digital sound the accuracy is only limited by the sampling frequency! (Theoretically giving us more then 20,000 sync-points in a second)

When actual sound work starts, the same process is applied to all sounds on the film. Transfer it to perforated tape, synchronize and mark it in the editing table and put the sounds on small rolls on the shelf. When all sounds are synchronized, they are categorized and assembled into tracks. Some tracks will have all the ambiances, some tracks have the footsteps, others have all the dialogue, etc. When building tracks the sounds are separated on the tracks by a blank leader-tape. This is to create space between the sounds. It is necessary for the sound technician to have space between the sounds so that he can adjust the filters and the sound-levels separately for each sound during the mix down. In the final mixdown there will be one track on each fader and there can be more than a hundred audio tracks going at the same time. One can spend months mixing a motion picture, a very expensive process.

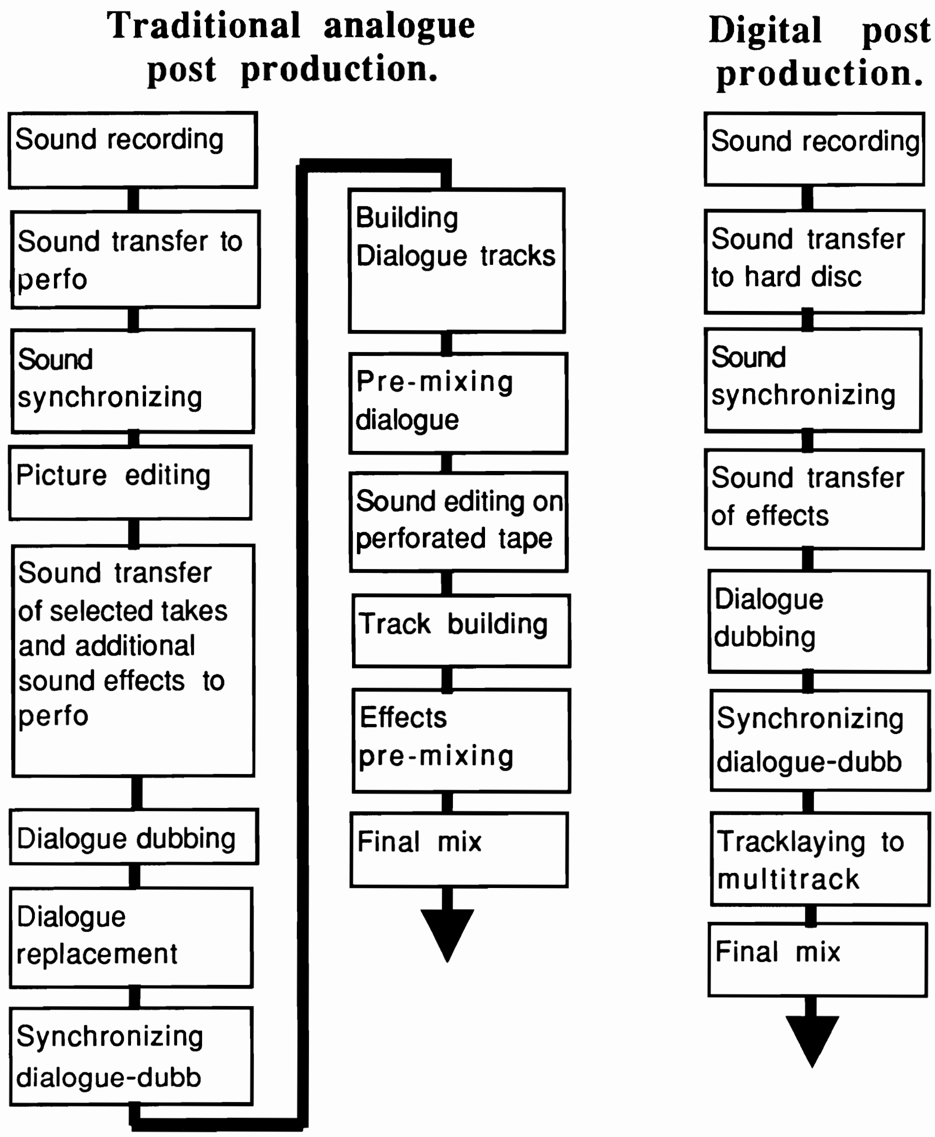


Figure 1. Phases of Audio Post Production.

Video & TV programs

Traditional sound work for video has been limited to the use of three to five audio tracks, depending on the equipment and the budget. **Using time in on-line is very expensive**.
In later years, as sound has become more and more important, the use of separate multi-track recorders has increased.

Multi-track / Synchronizer

For a multi-track audio recorder to run in sync with a video player, a synchronizer is required.

A synchronizer reads timecode from both a video player and a multi-track audio recorder. The synchronizer will then control your multi-track recorder and make it move in sync with the video so you can start laying tracks. You may want to think of the timecode as "electronic sprocket-holes".

CD-players

Sound-effects can come from another video player (controlled by the editor), or from any separate sound source that can be controlled by a synchronizer or an editor. If the sound-effects come from CDs, many CD-players can be triggered to start from an event-pulse generated by the editor or the synchronizer.

Track-laying

Track-laying on a synchronized multi-track machine is still the best solution for bigger productions. In spite of time-consumption and expense, most professionals prefer the advantages of going through a mixer that provides filtering and the addition of effects.

In traditional analog sound recording we have to consider the signal-to-noise ratio of the recording media and record everything at close to maximum level in order to avoid unnecessary noise on recordings. This means that loud sounds are recorded at the same level as soft sounds, thus making the sound-mixing a very long and demanding process. Several pre-mixes are often necessary in order to reduce the number of specific tracks.

Digital Post Production

Now, in digital audio post production things can be done a little differently. The signal-to-noise ratio is much better, and since we are recording numbers instead of actual sound, we can adjust the volume of any sound without changing the noise level.

Digital sound offers the ability to do track-jumping and pre-mixing without the traditional problems connected to analog sound recording, (hiss and treble build-up when doing generations). When the whole process is completely digital, there is no audible noise added.

When working with traditional analog track building, we are not given the possibility of fading sounds in and out during editing. All these fader movements have to be done during the pre mix or the final mix. Of course there are computerized mixing consoles that make this possible, but with digital sound, the fader movements are done during the actual editing.

The track building technique of traditional analog post production is unnecessary with digital sound since tracks are built as a part of the editing and synchronizing routine. These features speed up the editing and simplify the mixing process by removing the need for the often time-consuming and expensive pre-mixing.

In this chapter we shall see how digital sound recording and editing can speed up the post production process and leave more time for creative sound work. The introduction of **Studio 16 V2.0** and the **AD516** card, with the ability to do a mixdown from eight tracks into two in stereo, makes it possible to do the complete post production without going on tape.

(I would suggest a combination of a synchronized multi-track machine and Studio 16 for longer and more ambitious projects, mostly because I would want to do equalization and add reverb or effects during the final mixdown.)

1.3 PHILOSOPHY OF AUDIO POST PRODUCTION

Sound is part of our lives and, as filming has its roots in the observation of reality, one can't make films today without using sound. Silent films are history! Today the visual image is linked to sound. When you consider the visual, you must also consider the sound that will accompany it and the sound of every figurative element therein. Apart from dialogue the most important function of sound is to clarify what can't be explained through pictures. We spend our whole lives listening to the world around us, and building an inventory in our brain of not only the sounds, but consequently emotional circumstances associated with them. One should use this knowledge when constructing a soundtrack.

Make it Simple

When we are working with sound it is important to concentrate on the sounds that people can hear.

People can hear dialogue and music, or dialogue and special sound-effects at the same time. But if there are more than two basically different signals on the soundtrack, the audience will not be able take them in. Instead of clarifying the visual imagery, it will confuse the audience.

There is also another reason for not piling sounds, a technical reason; when you are putting one loud sound on top of another it doubles the energy and both sounds have to be played at a lower level in order not to exceed the maximum available output level. If you put the same two sounds after each other you will get a feeling the sound is louder, and the ear has a chance to distinguish between them.

Background Noise

One of the basic problems with dialogue-editing, is background noise. Background noise is the sound that is actually present in the surroundings we record in, and must not be confused with *atmosphere* or *ambiance*.

A dialogue recording done in a moving elevator will have a certain background noise - the sound of the elevator. This sound is very even and will not give us any trouble later when editing the dialog. A dialogue recording done out on the street will have a background noise that is not very even. The angle and the direction of the microphone will not be the same for all shots, and there will be different sounds coming from the traffic. This background noise will give us problem when we're editing.

The dialogue is there to be heard. The audience will listen for the dialogue no matter what the actors say or how they say it. But if a scene is cut where the background changes drastically, the audience will hear it and draw their attention to the editing itself. When the audience is starting to focus on technical events instead of the action, their concentration fades and they will soon go get some more popcorn or switch to another channel.

The best way to tackle this problem is to find the scene with the highest background noise and extend it over the cut to enable long cross-fades. That way discreet changes can be made over time and the differences in background noise are not noticeable.

Background editing is easy to do with the help of digital sound editing. With Studio 16 you can search for long periods of background noise between dialogue, and by copying them and pasting them repeatedly before and after the dialogue you can use scale to fade them in and out and make clean transitions between cuts. In fact, the trick in dialogue editing is to carry a proper background. Then you are free to do anything you like to the dialogue, (like changing lines and integrating dialogue-replacement) as long as you don't disturb the tonal quality of the scene.

A rule of thumb when you are shooting on video is to let the camera run for a few seconds after each take. That will leave you with plenty of background noise for later editing.

Ambiance

Ambiance or atmosphere is there to make the illusion perfect. When doing a scene from a boat in stormy weather we first showed the boat on the sea accompanied by the sound of the wind howling and the sea going high. But the dialogue scenes inside the captain's cabin, were shot either in a studio or when the boat is lying still at the pier. To create the illusion of a continuous storm outside, we can just add the sound of one! The ambiance is not necessarily the actual ambiance of the filmed location. The ambiance is what we add to give a scene the right feeling.

Perspective

Perspective can be achieved in different ways. When a voice becomes distant its characteristics change. It becomes thinner! The further away, the quieter it gets. The quieter it gets the less bass we hear due to the construction of our ears. This is why we have physiological low-frequency compensation on our home stereo sets - loudness.
(So we can adjust the volume and take out some of the low frequencies.) When we take the volume down on a sound track, we get another problem: How will the audience be able to hear it? Maybe they are sitting far away from the TV set or maybe the dishwasher is running in the background. You will have to make the audience accept the idea of perspective without changing the level too much. So cut the bass to give the illusion of the sound being further away, but we keep the level high, and maybe add a *little* touch of reverb or delay.

It is essential that the dialogue is heard, because, if anyone in the audience were to ask what was said, no one in the room could hear the next three lines and everyone would lose important information!

Dynamics

Good sound demands a good dynamic range. A good dynamic range demands a sophisticated high performance monitoring system. A TV-set cannot fulfill that, so we have to pull our sound together. That is, reduce the dynamics of the sound. Traditionally there are two ways of reducing the dynamics; one is *limitation* the other is *compression.*

Limitation is when you set a definite maximum level and prevent peaks to exceed that level by chopping them off. Heavy limitation makes the audio loose all its finesse and makes it sound squeezed. This is what we do with commercials when working with traditional tape recorders.

Compression is a much more gentle way to treat a sound signal. It only compresses the signal at an adjustable rate. A compression rate of 2 to 1 means that when your actual peak is 10dB above 0 the compressor takes it down to 5dB above 0. If the peak is 20dB above 0 the compressor takes it down to 10dB above 0. This does not ruin the sound in the same way that limitation does.

We will always have to use a certain amount of compression in order to get a dynamic range that we are able to handle. A little bit of compression at the lower levels holds it together. If the compressor is set up so that the break away point is 8dB above the operate point, there will be no compression on the normal levels, but as soon as the level gets above the operate point, the compressor pulls back very nicely without chopping off the top as a limiter would do, giving the sound room to expand.

*(When working with **Studio 16** the dynamics can be limited without making it sound squeezed. In the dialogue tracks, the fx-tracks or the final mix there are only relatively few places that need limitation. By using **Scale** to adjust the volume of the peaks only, the same loudness can be achieved as with heavy limitation, but without completely losing the open feeling. We will look at that later.)*

1.4 GATHERING SOUNDS

The most important skill for any sound editor is to develop the ability to create emotional sensations with sound. The experiences one gets from collecting sound often give lots of ideas about how to simulate emotionally the situations. In many situations one has to pick sounds that bear no relationship to the original event. These new sounds, when correctly combined, can still create the right feeling.

Using CD Archives

The easiest way to gather sounds is getting them from CD libraries or archives. There are quite a few CD archives around, and the sound quality is usually quite good. They are easy to use and often come with an index, both on paper and diskette. However, once you start working with CD sound archives, you will soon discover that there are only a few sound-effects that you use, and that you use the same sound-effects over and over again. My experience with CD archives is that most sound editors prefer the same sound-effects. I end up having the same baby crying, the same train passing in the distance, and the same wind through trees on our films as everyone else has on theirs. Even the audience starts recognizing the sound-effects. When using archives, use your imagination and the editing facilities of Studio 16 to prevent this.

Experimenting with sound is very funny and very easy with Studio 16, but be careful, it can be addictive!

Make Your Own Sound Archive

Of course the best is to create your own sound archive. My most recent method involves the use of Studio 16. When I am out recording sounds for film or TV programs, I always record many more sounds than I need. I also record a lot *longer* sounds than I need, just to be on the safe side. When recording ambiances, I always run the tape for at least 5 minutes. An ambiance recording can't be too long!

I record these sounds on a portable DAT recorder. And, when I get back to the studio, I transfer them to hard disk. Then I edit them using Studio 16 to cut off all unnecessary beginnings and endings. I always use at least 40,000Hz as sampling rate. Although you won't hear frequencies over 16 - 17KHz, the presence of high frequencies in the signal will interfere with the sound of the lower frequencies.

Storing On DAT

Once all the sounds are edited, I categorize them and transfer everything back to DAT cassettes using the DAT-recorders ID number in order to give each sound a number. The ID-numbers are essential in order to locate the sounds for later use. The ID-numbers go from 01 to 99. That means that there are only 99 locations available to store sound in. By using a mixture of both sound-effects and long ambiances on the same DAT, the chances of running out of ID numbers are small. Most modern DAT players have advanced locators and you can just punch in the number of the sound you want. It will locate and play back from exactly where you put your start ID. Some DAT players also have a small amount of RAM to store the first seconds of sound, thereby enabling instantaneous playback. (I usually push the start button on the DAT recorder half a second before I start playback from Studio 16. This way I can always be sure that beginning of a sound is recorded to the DAT.)

Cataloging

I use a text editor to make a sound index of my sound-effects. (Cygnus ED is by far the fastest!) I use the Amiga's multi-tasking capabilities to have both the sound-index and Studio 16 running at the same time. When I need a sound-effect from the library I press the left Amiga-button and M to switch to the index screen. Then I use the search option of the text editor to find the sound I need, and the information on where to find it. If it's on a DAT cassette, I insert the cassette in the DAT player and record it with Studio 16. Studio 16 can only play back sounds at one sampling rate at a time, so remember to check the sampling rate when you record to hard-disk. Use the same sampling frequency for the entire production.

Of course, if you use a very big hard-disk, or if your sound library is not very big, you can keep all your sounds on hard-disk. However, my experience is that keeping only project related sounds on the hard-disk makes the system faster and more flexible. It also gives you greater freedom in editing since scales and fades are permanent.

Mono - Stereo

Stereo transmissions have recently become more and more usual, but so far, the majority of the viewers are still watching programs on mono TV sets. When recording stereo sounds make sure that they are *mono compatible.* Our stereo recordings must sound good when played in mono as well as in stereo. Always check the mono compatibility of stereo sounds before including them in an archive in order to prevent future problems, and remember that a good monophonic sound will always sound good, both in stereo and in mono!

(A recommended stereo recording method is the MS-stereo technique, a 50 year old Danish invention. MS-stereo will always sound fine in mono.)

1.5 SMPTE / EBU - TIMECODE.

Understanding timecode is essential when working with film and video. The use of timecode is a necessity for everyone who wants to take advantage of Studio 16's excellent features for applying audio to film or video programs.

History

Although timecode has only come into common usage in the relatively recent past, the history of its development spans more than thirty years. In traditional 35 mm motion picture film it's easy to locate the appropriate cutting points by examining the film and using the sprocket-holes as a reliable reference to identify the edit point. In video; however, there are no sprocket-holes, nor can visual examination give any indication of the information recorded there. Since video is recorded diagonally across the tape it cannot be edited by the normal cut and splice method used in film and sound tape editing. In 1967 the first steps towards a digital representation of the time position of each video frame were taken, and in 1972 the Society of Motion Picture and Television Engineers(SMPTE) and the European Broadcast Union adopted the idea and established the timecode standards we are using today, (SMPTE/EBU time code).

LTC

The timecode most commonly used today is called LTC (Longitudinal Time Code) and is recorded continuously along the length of the audio track of a tape, thus the name. The timecode allows you to identify each video picture individually by assigning a time value to each frame and encoding this on the tape. The information recorded on each frame consists of 80 information units, each being set to either 0 or 1. One frame of timecode consists of 32 user bits, 16 sync bits, 31 assigned address bits (hrs.mins.secs.frames). + one unassigned bit (bit 58). The initial thrust behind the establishment of time code standards, lay in the editing of video tapes, and this function still remains central to the application of SMPTE/EBU code. Today numerous synchronization systems offer an inexpensive and reliable way of synchronizing audio and video equipment. Additionally, timecode can be used to accomplish synchronization between video or audio tape transports and MIDI (Musical Instruments Digital Interface), an area that has become very popular.

VITC

As you have seen earlier in this manual, there is also another timecode format, called VITC, (Vertical Interval Time Code). This code, inserted in the gap between the visible frames (the vertical blanking interval), offers the advantage of being legible under any circumstances where a picture is available. Although well suited in video editing where much time is spent in slow-motion and still-frame modes, VITC has some major limitations. It cannot be used with audio recorders without a VITC to LTC converter and can only be written on tape simultaneously when recording the video signal, as the two share the same recording channel. Some digital audio editing systems let you scrub the sound in search-mode. Since LTC is not reliable when being read at very slow speed, VITC is a necessity when working with these systems.

Recording Timecode

A timecode signal recorded on an audio track alternates roughly between 1 and 2 KHz, which is right in the middle of the frequency range of an average audio recording channel. LTC is remarkably reliable on today's video and audio recorders and will rarely cause any trouble. Studio 16 is capable of reading relatively bad timecode and is not stopped by occasional drop-outs. Regenerate or refreshen timecode when copying it to another audio channel. However, there is one thing that can seriously damage the intelligibility of any timecode. That is the use of a noise reduction system, e.g., Dolby or dbx. In the professional world of video and audio it is a rule of thumb to disable noise reduction on any channel used for timecode recording.

Striping Audio Tape

Professional setups will normally include a **Master recorder** onto which either the final mix down is done, or the final mix is transferred. The reason for this is that high quality video is normally mastered on very expensive high quality VTRs only found in on-line video-editing suites. The audio-master is taken there to be transferred to the video master. Despite the fact that most of you are editing the audio directly to the video master, the process of going to an audio master tape and then taking this to an on-line suite for layback, is worth a minute of discussion.

Using a 1/4" Analog Tape Recorder

The 1/4" tape recorder is still the most used audio master format. Ordinary 1/4" tape recorders are found in most places where professional audio is done. Normally they have two tracks, but the last few years a center-track-rack for timecode has become more and more common, (CTTC). When doing audio layback it is essential that the sound is synchronized with the video. There are two ways of obtaining sync. One is to record the timecode from the VTR onto the center-track (or one of the audio tracks) at the same time as recording the mix onto a free audio track (remember to regenerate). The other is to pre-stripe the audio master with SMPTE and use a synchronizer to keep it in sync with the video when either doing a mix down or copying the mix to the audio tracks. The first method is the best if the final mix down is made internally using Studio 16. The second method is the only method that allows you to do punch-ins during mixdown.

Record timecode at 10dB below 0 VU!

Using a DAT Recorder

If the audio master is a DAT-tape, there is no need for striping it with timecode. Most DAT recorders leave a very reliable pseudo timecode called A-code. This code can easily be translated into SMPTE by the professional DAT players often found in on-line suites and video facility houses. In some cases the DAT master is re-recorded on a 1 C-format tape, or on a BetaCam SP tape and synchronized by the editor before layback.

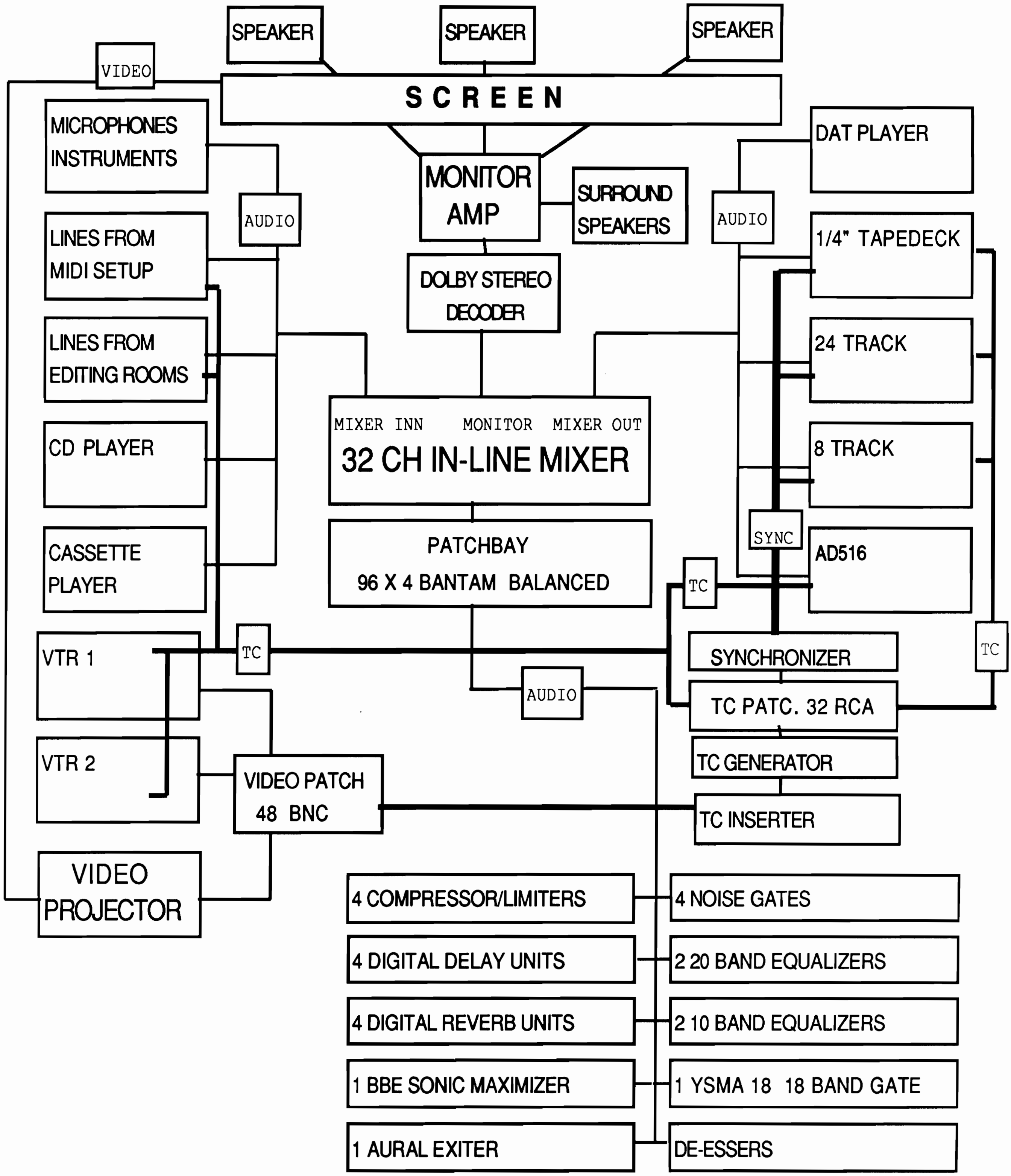


Figure 2. The Sound Factory Configuration.

2. AUDIO FOR VIDEO USING STUDIO 16

2.1 USING STUDIO 16 AT THE SOUND FACTORY

In today's professional audio post production studios, digital audio has already proven its advantages and has become by far the most popular editing tool. Digital audio editing is applicable to almost any task normally performed in audio post production. In **Figure 2** you can see the configuration at **The Sound Factory's** mixing studio. **The Sound Factory** is a professional audio post production company with sophisticated equipment and high standards. Our customers come mainly from the film industry but also from the video and music industry.

Films are scanned to U-matic Highband prior to the sound work. We use a Sony editor to control the VTR and a combination of Fostex and Tascam synchronizers to control the tape decks. The picture is projected on a perforated screen 4.5 by 3.5 meters. A Dolby Stereo system with 3 Klipsch theater speakers behind the screen each driven by an Electrocompaniet amplifier provides good listening. Studio 16 plays an important role in both our two sound editing suites and in the mixing theater.

Since October 1991 the dialogue and background editing of two major Norwegian motion pictures has been done using Studio 16, and during 1992 the first complete Dolby Stereo soundtrack will be completed with the sole use of Studio 16 /AD516

Working methods

When working with feature films, the sound is transferred to hard-disk, edited, synchronized to video, and then transferred to a multi-track recorder. Simple soundtracks for short films and documentaries are done without going on multi-track. Although this setup may look impressive and expensive, the major differences between this setup and the budget setup in figure 3, are the patch-bays and the multi-track recorders. The patch bays are what really makes the setup flexible. The audio patch, the video patch, and the timecode patch are essentials in a professional post production studio. A well-planned audio patchbay will do wonders to any professional sound studio, and a separate timecode patchbay can prevent a lot of problems. A little stray timecode can go a long way, and it usually goes to where you absolutely don't want it, (on your dialogue tracks).

Working with long films requires careful planning. The tracksheets are often written before we have recorded a single sound! Even with 24 tracks it's easy to run out of tracks if there is a lot of dialogue and post synchronized effects. This is where Studio 16 can really save a lot of time and frustration. All effects can be edited and premixed digitally using Studio 16's Transport module. Transport allows recording from the Cue List, which makes it possible to build very complex effects tracks without running out of tracks on the multi-track.

2.2 THE BUDGET SETUP

Integrating Studio 16 in a video editing setup

Many video companies already make use of the Amiga's power in graphics and titling.
To include Studio 16 is not a big expense, since the rest of the stuff is already there.

In professional video production the original recordings are done with a BetaCam SP recorder or a portable 1C Ampex-Nagra. Then all the material is transferred to VHS or U-matic for editing. This editing process is called off-line editing. Off-line editing equipment is not broadcast quality and therefore much cheaper to work with. This is important since the editing process normally takes a long time. After the off-lines edits are completed, the final edits are done on-line in the studio.

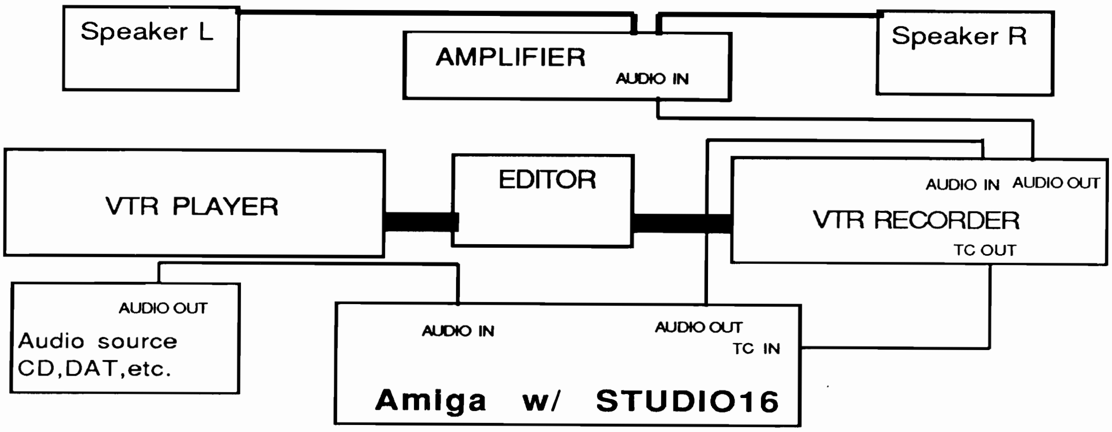


Figure 3. Integrating Studio 16 in an Existing Setup.

Incorporating Studio 16 in an off-line editing setup is an affordable solution with many advantages. With Studio 16 it's easy to try out both sound-effects and music as well as voice-overs (narration by a speaker who is not seen) during the off-line editing, which is very helpful to both the director and the editor.

Example: Editing voice-over

1. Transfer the voice-over to hard-disk.
2. Edit the voice-over to the off-line-master and save the Cue List.
3. The voice-over selections are now done, and the unnecessary takes can be removed to save hard-disk space.
4. When the off-line editing is done, the video is on-line edited in an on-line editing suite where you edit from the originals to a 1C-format or BetaCam master tape.
5. After on-lining the video master, all you have to do is find the new in-points.

Contrary to traditional working methods, where the voice-over is transferred to VHS or U-matic and edited with the off-line equipment, Studio 16 remembers all the edits and the voice-over is ready to be transferred to the video master. The professional producer knows how much time (and money) this will save! (On a documentary program 25 minutes long the voice-over editing is between 3 and 6 hours work.)

The same can be done with music and sound effects. One often comes across great ways of doing sound-effects when playing with sound in the off-line room that one is never able to repeat when the actual sound-editing starts. With Studio 16 none of those great effects are lost!

The following is a good setup for semi-professional audio for video, suitable for TV-stations and small video production companies, but also a good setup for a freelancer who wants a mobile solution. Let's have a closer look at the configuration:

The VTR player

The purpose of the VTR is playing back the picture we are working with, and at the same time feed timecode to our AD516/AD1012 card as well as any synchronizers or midi-equipment that is connected.

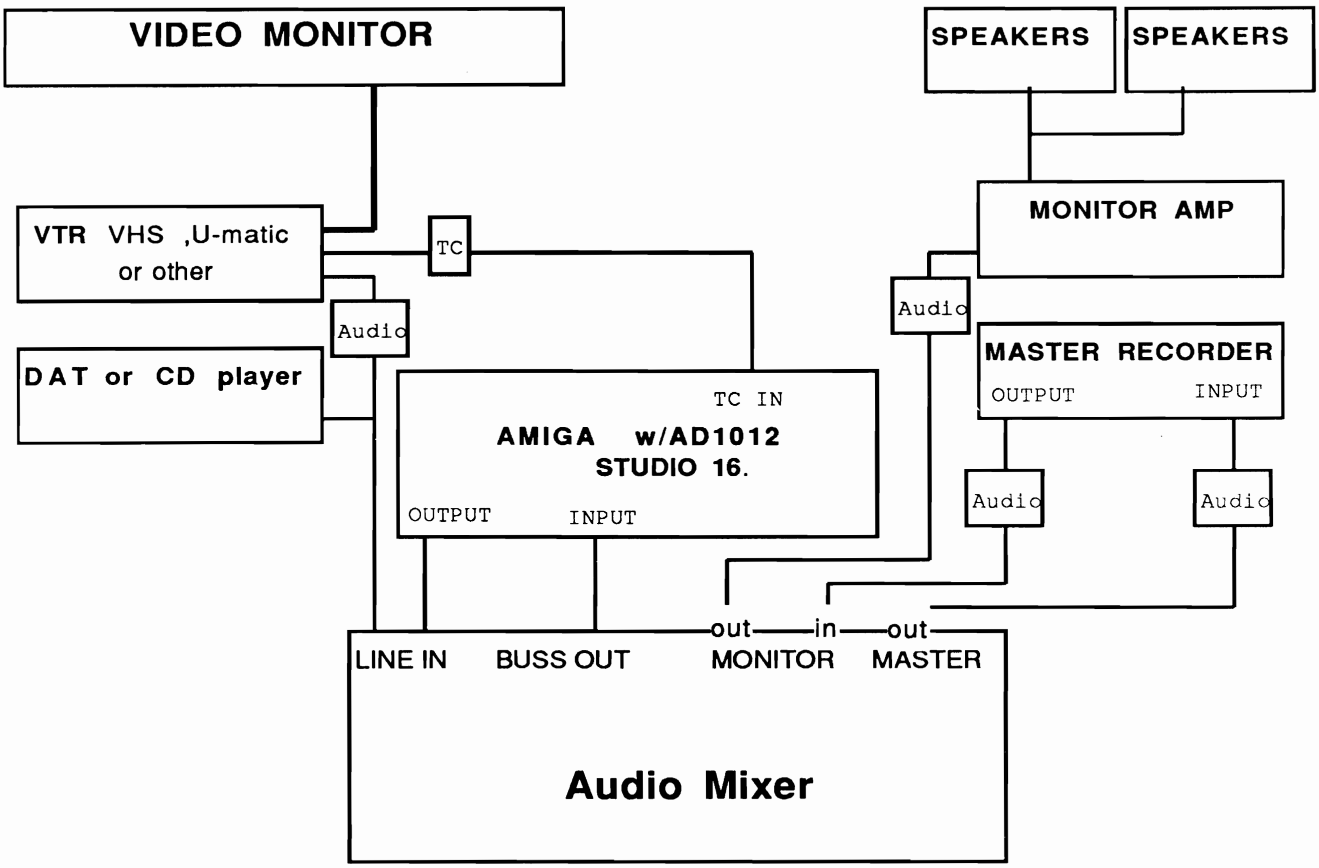


Figure 4. The Semi-Professional Setup.

The video tape recorder can be anything from a semi-professional VHS to a professional 1C reel to reel. The best solution is probably something in between like a S-VHS or a U-matic Highband SP. These players have a separate timecode channel that, as discussed later, makes life a bit easier when we are editing and synchronizing sound. It is highly recommended to use a VTR that has a shuttle wheel. If not, it might be very difficult and time-consuming to locate exact editing points.

The sound source (CD-player, DAT-player or other)

The purpose of the audio source is to feed sounds to the AD516/AD1012 for editing and synchronizing with your video program. A CD-player, a DAT-player or a professional 1/4" tape deck are preferred as audio sources, by audio technicians, but if you find the sound quality acceptable you can of course use any media you prefer. If you are using sound libraries you will find that they are most often distributed on CDs. A lot of these are of excellent sound quality, but there are in fact some that are converted to digital from old analog recordings.

Some of the older archives (like Sound Ideas 1000 series) contain mono sound effects that are processed for artificial stereo. This process makes them sound funny in mono, (very low mono compatibility).

Make sure that you record only one channel if you have to use these sounds. They don't come out well on a mono TV set. Another thing about CD archives is that viewers start to recognize the different sound-effects, as the same sounds are often used in several programs.

The Audio Mixer

The purpose of the audio mixer is to make it easy to feed signals to the sound recording media and to monitor the sound. The mixer is not an absolute necessity. Many beginners don't want to spend their money on a mixer. (Instead of a mixer you can use a patch bay.) In this manual; however, I will assume that you are using a mixer.

It is important that the mixer is flexible and quiet. Flexible means that it is easy to perform different tasks like copying any audio source to any recorder and being able to switch back and forth between the original audio source and the playback from the recorder while recording (AB-test), thus enabling you to make a direct comparison between the original and a copy.

Quiet means that the mixer does not produce a hiss of its own so loud that it exceeds the signal-to-noise ratio of your recording media. For AD516 that means a SNR of about 90dB. When all the mixer's input channels are set to the line in position with the faders in the nominal area, it should not have a signal-to-noise ratio lower than 90dB with a signal being fed to one line input.

Other important mixer features are that it has a good filter bank for equalization, an output section with at least two recording busses or group outs, and a monitor section that allows you to adjust your listening level without adjusting the master level. Choose a mixer with long faders. Short faders will not feel comfortable and will make it difficult to carry out small adjustments in volume at low levels, (e.g., when balancing backgrounds and ambiances)

Multi-track recorders

The time has passed when multi-tracks were expensive machines only to be found at expensive recording studios. Today a multi-track is within reach of most video production companies' budgets. Presently an eight track analog multi-track can be purchased for roughly $2,500; whereas, a 1" professional 24 track will cost $10,000 - $12,000. The ADAT, a digital 8 track DAT recorder from Alesis, sells for $4,500 and is a good partner to Studio 16.

The Master Recorder - VTR or audio tape deck

The master recorder is where one puts the final mix. Whether it's a VTR or an audio tape recorder, the setup will be the same. If you are working on a beginner's or semi-professional's level, you will probably do the audio layback direct to the video masters audio channels (Audio dub). If your VTR has a timecode channel, just hook up the timecode channel's output to the AD516/AD1012 and record the soundtrack onto the video masters audio channels.

If you are a professional you will do audio layback differently. The master recorder will normally be a professional audio tape deck or a DAT-recorder, and the video master will be a BetaCam SP or a 1C-format tape. The final mix will be recorded either on a DAT or on an ordinary 1/4" tape and taken to a video on-line editing suite, where the final mix will be synchronized with the video master and transferred to the audio tracks.

The audio amplifier and the audio monitoring system

A chain is only as strong as its weakest link! Unfortunately the weakest link in audio post production is very often the monitoring system. A good monitoring system is Alpha and Omega when you are working with sound. There is no point in working for days to make your program sound right only to find that when it is shown on the TV it sounds so wrong that you wish you had never been born. Some of you, like me, may have experienced that already. This happens when the monitoring is not good enough. What you hear is your only guide no matter what!

Although you will get used to your monitors after a while and will be able to compensate, imagine telling a big shot producer who wants to be impressed: "Ahh, don't worry about the sound..ah, er, it will sound OK on air! You see, my monitoring is......".
When working with audio for video a 'neutral close monitoring setup' will always be safe. Neutral means that it does not color the sound, i.e., it does not favor or disfavor any part of the frequency range. Close-monitoring means that your speakers are designed for listening at close range, (3 - 5 feet). These systems are not very expensive and there are several good systems around. Ask local professionals what they use.

2.3 AUDIO MONITORING

Adjusting the monitoring

To obtain a neutral sound from your monitor system it is necessary to adjust it. Use a pink noise generator, a frequency analyzer, and an equalizer.

1. Plug the pink noise generator into the frequency analyzer and look at the graph. Some analyzers can memorize several readings. (If this is the case with your analyzer, you should store the graph from the pink noise generator.) If your analyzer is unable to store data you will see that the pink noise generator is producing an even graph.
2. Plug the pink noise generator into your mixer. (Preferably into a tape return socket where you cannot have the filters working by accident.)
3. Insert the equalizer between the mixer and your monitor amplifier.
4. Turn on the pink noise generator and use the frequency analyzer to measure the sound coming from your speakers. The analyzer's microphone must be located at your listening position and pointing at the speakers!
5. Compare the graphs.
6. Adjust the equalizer until the graph from your speakers are even with the graph from the pink noise generator.

Your monitoring system should be fine now. If you think it sounds dull, remember: this is a reference system and not a hi-fi stereo set. If you're in doubt, always consult someone professional on monitoring.

Work in quiet surroundings

Avoid heavy noisemakers in the studio. If possible keep the Amiga and the VTR in another room. The noise produced by their cooling fans will interfere with the listening and might remove attention from unwanted changes in the background noise of the recordings. Changes that would otherwise be clearly audible when presented on air, and to people listening in quieter surroundings. Unnecessary noise from the surroundings will also ruin your concentration and prove exhausting.

3. TRICKS & HINTS DEPARTMENT

3.1 EDITING AND SYNCHRONIZING DIALOGUE

Picture and sound are traditionally recorded separately. With video came the ability to record both sound and picture on the same media. This has solved some of the problems with sound (it will always be totally in sync with the picture), but it has also created some new ones. With the cameraman who now decides when to record both picture and sound, the sound editor is often left with far too little both at the beginning and at the end of a scene.

In the Stone Age the director used to shout: "Run sound!" and then soundman ran the tape recorder and, when he was ready, shouted back: "Running!". Which was followed by "Camera!" and "Action!" before the actual shooting of the scene started. This left the sound editor with lots of background noise that is badly needed when working with dialogue. In dialogue editing we are faced with background noise levels moving up and down in every cut. Studio 16 offers solutions to these problems.

Preparations

By the time the editing of the film is done it is time for the audio post production to start. The editor has now decided which takes to use and has produced a video master with a very rough edit of the original sound on one or two tracks. The master must have timecode recorded on one of the audio channels or on a separate timecode channel.

You should always work from a copy of your video master when you are editing sound!
Record the timecode on the timecode channel, or on one of the audio channels, and do a window dub using Studio 16, the SMPTE Monitor window, and a genlock.

(*If you don't have a genlock use a timecode-inserter. A timecode-inserter will also read the user-bits!*)

On the other audio channel(s) the rough edit of the sound must be recorded in order to have an accurate sync-reference for the dialogue re-synchronization. Then all the dialogue scenes must be transferred to hard-disk. Use a reasonably high sampling rate.
Although many VTRs don't offer a frequency range better than 30Hz to 15 KHz, use at least 40,000 Hz (44.1KHz on 16bits).

An example - A dialogue between two people:

First Shot - A wide shot of Tom & Fred in the kitchen. There is a coffee machine working in the background. Tom is sitting at the table. Fred is opening the refrigerator and taking out a bottle of milk.
Tom: "*What's for breakfast?*"

Second Shot - Close up of Fred looking at Tom.
Fred: "*Have you called your mother?*"

Third Shot - A close up of Tom looking down at the table. Suddenly he looks up.

Fourth Shot - A wide shot of the kitchen. Fred has closed the door of the refrigerator with a bang and is walking over to the table. He pulls out a chair. As he sits down he says...
Fred: "*Have you called your mother yet?*"

The chair is dragged to the table.

Editing dialogue

The first shot is wide and has rather loud background noise. It also contains sounds from the refrigerator's door being opened. We are going to concentrate on getting the dialogue sounding its best and at the same time make a neat transition over the next cut and into the next dialogue.

First, we cut out all the sound that contains unwanted elements at the beginning and at the end of the samples.

When shooting a film, it sometimes gives a better result to record the dialogue on a separate DAT recorder compared to recording it on the audio channels of the video. However, this procedure presents a problem. We have to get the sound synchronized to the picture.

Using Studio 16, the next step is to find the right start time.

1. Locate the exact starting point of the clapper or the first word of the dialogue on the video using the shuttle wheel. When you've found it, read the SMPTE off the video.
2. Open the Editor with the sound file and locate the clapper and mark it by clicking at the exact start point with the mouse.
3. Select Set Display Offset from the Options Menu. Enter the SMPTE from the video in as the Range Start number.
4. Now, the new number for Sample Start will be used as your start time in the Cue List.
5. Open Cue List and enter the sound file and the above Sample Start in the Start Time field.
7. Play the video and listen to both the VTR audio and the sample being played from the **Cue List**.
8. If they are not exactly in sync, try using the pan pots on the mixer to have the VTR's audio and Studio 16 playing in different speakers. That way it's easier to define which sound starts first. Adjust the start time in the **Cue List** until they are exactly in sync. Repeat the process for the next sample. This may sound a bit complicated, but it is really fast when you get the hang of it! (Future versions of Studio 16 might come up with an easier way to do this)
9. Now the replaced dialogue is playing in sync with the video, and you should turn off the volume of the video.
10. Save the **Cue List**!
11. Open the Editor again. Select Display Offset from the Options Menu. Enter the Cue List's start time as the Sample Start number. The **Status display** will now have the same reference as the video.
12. Look at the video to determine where the scene starts, range the part of the sample that precedes the start of the scene and use **Effects - Zero** to silence it.
13. Allow two frames of audio to overlap from the previous scene or make a quick fade-in from roughly five or six frames before the scene starts. That makes a smoother transition.

If the dialogue exceeds the dynamic range of the media, we can use **scale** to lower the peaks and lift the lower areas.

*Remember that **scale** is a **destructive edit** and will alter the sample permanently. If you lift any part over the top, it is ruined and you have to re-record the sample and start all over again. (I always make a copy of the sample before doing edits that I'm not certain about the outcome of.)*

The second shot as a close up allows a closer microphone, thus less background noise.
We have to avoid drawing attention to the cut. This can be done by letting the background noise of the first cut continue over the next. But, since the background noise from the last cut is not long enough, we have to extend it. In order to get a background noise long enough, we have to make a background noise track.

Making a background track

1. Try to find a passage of the sample that has no dialogue or other recognizable sounds in it.
2. Range it and select a **Destructive Copy,** so that a CopyBuffer sample is created. Rename CopyBuffer to BG1.
3. We have two alternatives for making a background track. One is to use the **loop** function:
   - Open an Editor for **BG1** and select Range All.
   - Activate Loop in the Options Menu and play the sample.
   - Adjust the length of the range until the loop-points so the sample is playing without noticeable transitions. Record it as a new sample using **Transport.**
   - Sometimes it is impossible to make a good loop.
4. We can use the **Cue List** to accomplish a crossfade loop.
5. Open an **Editor** for **BG1** and use **scale** to make a 25 frames fade-in and a 25 frames fade-out. Clear the entries in the **Cue List** and enter a start time of **00:00:01:00** for **BG1** .

START TIME	END TIME	VOL	SAMPLE NAME
00:00:01:00	00:00:06:15	+0	BG1
00:00:05:15	00:00:10:05	+0	BG1
00:00:09:05	00:00:14:20	+0	BG1
00:00:13:20	00:00:19:10	+0	BG1

Figure 5. Example of making a background track from the Cue List.

6. Duplicate the entry and look at the end time of **BG1** in the Cue List. (Figure 5)
7. Start the next **BG1** 1 second before the end time of the previous entry and repeat until you have a total time that covers the whole scene.
8. Remember to turn off the **Monitor** on the mixer window. It will add noise to your recordings.
9. Open **Transport** and the **SMPTE generator** and select Record Output from the Transport Menu.
10. Select Record and Play, and run the timecode generator. If you don't specify a name before recording, the sample is called Untitled.
11. Listen to the sample to hear if it's OK.
12. Rename it from The **open list** to **Background1**. (Save the **Cue List** just in case...)
13. Reload the dialogue Cue List and enter **Background1** from the beginning of the scene.
14. Open an **edit window** for **Background1** and enter the start time from the Cue List as SMPTE offset. Use scale to bring the volume of **Background1** up or down to make discreet transitions between the cuts.

An example of advanced dialogue editing

There are film directors in this world who wants the sound editor to perform impossible stunts! With Studio 16 the limits of what is possible are drawn even beyond a film director's imagination!

In the Tom & Fred example Tom asks, "What's for breakfast?". Unfortunately, Tom has a slight speech defect. He lisps. Especially when he is nervous. In the take selected by the director and used in the film he has a serious lisp. In one of the run-throughs before the actual shooting started, I made a test recording that was okay. The director wants me to exchange the words "what's" and "breakfast" using the words from the run-through. I enter both the **1.2 #6** (which is the name of the take actually used in the film) and **Dia rep** (which is short for Dialogue repair, and contains the recorded material from the run-through) in separate edit windows. I range the word "what's" in both **Dia 1** and in **Dia rep**. It's 17 frames long in **Dia 1** and 18 frames long in **Dia rep**. I range the word. I do a Destructive Copy of the range in **Dia rep**. Then I extend the beginning of the range in **Dia 1** so that it reads 18 frames. Then I select paste, replace, and OK. When I play it back we both get a laugh. The two 'whats" are so different in pitch and sound that they are quite un-exchangeable. It sounds strange and would totally ruin the scene if used.

Plan B - Could we change just the 's'? Why not give it a try? Using zoom I am able to find exactly where the 's' is and range it. Both in **Dia 1** and in **Dia rep** I make sure to include the end of the t as well, and also the beginning of the f that immediately follows. They are both the same length (4 frames) so I do a Destructive Copy of the range from **Dia rep** and paste it with the replace option to **Dia 1**. It works perfectly! The director is very pleased. I do the same with breakfast and the problem is solved. Imagine doing that with traditional analog cut-and-splice editing!

Premixing the dialogue (on a four track system)

We end up doing a premix of the dialogue and the background by making one track for each scene of the film. Since all internal mixing is digital, there is no significant loss of quality. When all the scenes are done, we can do another premix. This time we will mix all the dialogue tracks and end up with only one dialogue track for the entire film.

(If you are using the AD516 there is not much advantage from this last premix)

3.2 POST SYNCHRONIZATION OF EFFECTS

When the soundman is recording on the set, naturally his mission is to pick up the dialogue as well as he can. Since recording dialogue requires the use of directional microphones, other important sounds will often be off mic. They will sound muffled and dull. These sounds have to be added during the post production. If the soundman has made wild recordings on the set, that is easiest. If not, we have to either look in a CD archive, or preferably record them from scratch. Some bigger facility houses have dubbing theaters for film, where one can make all sorts of effects while viewing the picture. For video programs, with the usual low budget, it is just as well to record wild in the real world.

Making a track sheet

In the little scene with Tom and Fred a lot of sound-effects are needed. The sound editor and the film director agree on which sound-effects will add the desired feeling, and the sound editor can start planning the tracks.

Spend time organizing the sound-effects tracks. Careful planning is essential if you are trying to do the entire film digitally with Studio 16. If the sounds on this tracksheet are premixed, it will be difficult to change levels on specific effect later as the sound of the coffee machine is constantly playing and will consequently change level too.

Effects

Editing sound-effects is different from editing dialogue. When editing sound effects we isolate the effects completely from the background noise and try to get them as clean as possible. Use quick fade-ins and fade-outs (1-5 frames) to ease them in and out. Not all sounds can be used in their original form. Sometimes it is necessary to run them through equalizers, exciters or enhancers. The ability to treat specific sounds with different outboard equipment is very limited since neighboring sounds may be affected un-intentionally. Apply the necessary processing to the sound before going on hard-disk. This method offers the ability to hear the effects with the rest of the sounds (dialogue, ambiances and music), and at the same time, to go back and change the level of any sound-effect at any state of the post production (provided the planning is good enough).

Studio 16 can playback eight tracks simultaneously with a reasonably fast hard-disk on an accelerated Amiga 2000 or an Amiga 3000. Compose your tracks so that the sound-effects do not overlap on the same track. You can have eight premixes playing at the same time, so there should be no problem avoiding overlapping.

TRACK SHEET

TIME / SCEENE	TRACK 1	TRACK 2	TRACK 3	TRACK 4
00:02:20 WIDE SHOT TOM AND FRED KITCHEN	REFRIG. DOOR RATTLES SQEEK	CHAIR MOVING	COFFE-MACHINE.	
00:08:23 CLOSE UP FRED	BOTTLES MOVING IN FRIDGE		LOUD HISSING	
00:11:11 COSE UP TOM LISTENING		REFRIGE-RATOR DOOR CLOSING	FADING OUT SLOWLY	
00:16:07 WIDE SHOT FRED COMING TOWARDS CAMERA. SITS DOWN	FRED WALKING TOWARDS CAMERA FRED SITS DOWN	FRED PULLS CHAIR		

Figure 6. Track Sheet for a Four-Track System Leaving Track Four Free Just in Case of a Premix .

Using the Cue List

Synchronizing effects is very easy with Studio 16. Locate the sound start on the video and enter the start time in the **Cue List.** (Figure 7)

START TIME	END TIME	VOL	SAMPLE NAME
00:00:02:20	00:00:05:11	+0	Refrigerator Door opens
00:00:08:00	00:00:11:16	+0	Bottle1
00:00:14:07	00:00:17:11	+0	Foots Fred
00:00:19:16	00:00:21:07	+0	Sits down

Figure 7. Example of Cue List. FX1.

Look at Figure 6. That's how it was planned. Then look at Figure 7. This is how it turned out.

Sound aid

Notice that the sound of the bottle inside the refrigerator is starting right before the cut. This is an example of how sound can help softening a cut. If the audience is presented with a sound, they want to see what caused it and will accept almost any cut as long as they are rewarded by seeing where the sound came from.

In **Figure 8** we have an example of another way to use sound as a helping element.
At 11:11 the sound Refrigerator door closing starts. The reason for this sound to start at 11:11 is that at 11:17 Tom looks up from the table. The sound helps motivate Toms eye movement, like the eye movement initiates the cut in the video.

START TIME	END TIME	VOL	SAMPLE NAME
00:00:04:10	00:00:07:09	+0	Chair TOM
00:00:11:11	00:00:14:00	+0	refrigerator door closing
00:00:17:12	00:00:19:15	+0	Pulling chair

Figure 8. Cue List track 2 FX2.

At 10:15 the coffee machine is giving a loud hissing sound like coffee machines do when they have almost finished making coffee. The reason for adding this sound-effect and putting it there is that Fred has just finished his line, and the hissing sound adds some sort of punch to it. (If we used the sound of an ambulance passing outside instead of the hissing sound, we could give the audience associations to hospitals, and create a different feeling in the scene. Imagine what the sound of a distant circus or machine guns would do.)

Using a sync pip

The purpose of a sync-pip is to help synchronizing an audio tape to a video tape. Record a 1KHz test tone for one second. Studio 16 is able to produce one from the Editor (Gen Sine). Make it exactly one frame long, and call it PIP. Use this pip for sync-checking by entering it in the Cue List at exactly 5 seconds before the video starts.

(On film this pip is entered exactly on the last clean number of the leader, 48 frames before the picture starts, but all you guys working with film know this already.)

For video, 5 seconds seems to become the standard length If our film starts at 00:00:02:20 we must enter a pip at 23:59:57:20 in order to get it 5 seconds before first picture. Some synchronizers have trouble figuring out the offset when they are passing midnight.

(A professional video-program or a TV-transmission master for TV normally starts at 00:02:00:00 or 05:00:00:00)

If you want to do the audio post production completely digitally with Studio 16, you can use **Transport** to mix FX1 and FX2 into a single FX track. The **Cue List** will look like this:

START TIME	END TIME	VOL	SAMPLE NAME
23:59:57:20	23:59:57:21	+0	PIP
00:00:02:20	00:14:32:20	+0	FX1
00:00:02:20	00:14:32:20	+0	FX2

Figure 9. Mix Down Cue List to a single FX Track.

Name the pre-mix **FX track.** The ability to change levels still exists but is greatly reduced due to the overlapping sound-effects.

3.3 AMBIANCES

Some sound editors add ambiance to mask bad dialogue or bad background editing. Used like that, ambiances can make the soundtrack seem untidy and it can be tiresome to the audience. The purpose of using ambiances must be to add the right feeling to a scene.

If we want our dialogue scene with Tom and Fred to take place on a Sunday morning, we can add a touch of distant church-bells. If we want them to live by the sea, we add some seagulls.

Be careful with ambiances

Try to make ambiances transparent. Don't let the *energy* of the ambiance be concentrated in the frequency area where the energy of the dialogue lies. Often it is sufficient to add the ambiance only in the beginning of the scene and slowly ease it out when the mood is established. We are able to select which sounds we want to focus our attention on as long as we are talking about sounds that are present in our surroundings. However, if the sound is coming from a loudspeaker our ears will hear this as *one source* it can choose *either* to ignore *or* to listen to. Selecting what sounds to listen to *within* one source, is not possible.

If adding ambiance makes the dialogue suffer, leave it. It will only mess up the sound and wear out the audience.

Creating ambiances

Sometimes one is not able to find the right ambiance on CD archives or in real life. Studio 16 is the perfect tool for tailoring ambiances. The following **Cue List** gave the perfect ambiance for the opening of a short film.

START TIME	END TIME	VOL	SAMPLE NAME
00:00:01:00	00:04:00:00	-5	Sheep outdoors
00:00:01:00	00:04:00:11	-10	Cowbells outdoors
00:00:00:00	00:05:00:00	+0	Harbor at night, wood creaking
00:00:01:00	00:04:00:00	+0	Wind through leaves

Figure 10. Example of making a background track from the Cue List.

This was recorded on a DAT recorder and dumped back to Studio 16. Then the files from the Cue List were erased. 17 minutes of sound takes up a lot of space and by doing this pre mix I saved 13 minutes, (65Meg). Ambiances are often very complex sounds with a wide frequency range, and must sound as good as possible, use high sampling rates!

3.4 VOICE-OVER (narration by a speaker who is not seen)

Voice-overs are commonly used in documentaries, reports, and commercials. Studio 16 is the perfect tool for editing voice overs. I often record voice overs directly on hard-disk, (with a DAT running as a backup). If any director, actor, or client wants to hear a passage played back, they can hear it immediately, just by the click of a button. I let Studio 16 take care of numbering the takes. I just write down the numbers as they are recorded, and the selection and editing is done very quickly. Careful compression is recommended when recording voice-overs. If the dynamics get too high, use **scale** rather than a limiter to control the peaks.

3.5 TRACK-LAYING ON MULTI-TRACK USING STUDIO 16

When a synchronizer and a multi-track tape recorder are used, flexibility is increased, and the process will be different. Using an eight track recorder (or 16, or 24 tracks) enables the use of several tracks of stereophonic ambiances and music without premixing, thus fixing the levels.

When working with a multi-track, some of the premixes become unnecessary. We can transfer the Cue List directly onto the tracks of the multi-track recorder. Avoid overlapping sound-effects, if you are pre-mixing before recording on the multi-track. If we should get second thoughts about the volume of some of the effects later, we can load the right **Cue List** and edit the volume of that particular effect and re-record the part of the track that contains the changes.

Sounds that need special treatment by equalizers, enhancers, etc., should be put on separate tracks on the multi-track. If there are not enough tracks, the sounds should be processed before being sampled into Studio 16.

Notice!
Some VHS players do not run steady enough to serve as master in a sync system, especially when recording music. Some synchronizers can compensate for wow and flutter on the master. If your synchronizer doesn't, you must connect the SMPTE directly from the multi-track's SMPTE out to the AD156/AD1012 SMPTE in, and disable the synchronizer while recording.

Be sure to save your Cue Lists even after you have laid the tracks, and keep them stored until the final mix is laid back to the master.

3.6 FINAL MIXDOWN USING STUDIO 16

Studio 16 can play a maximum of 8 tracks simultaneously. We have nine tracks to mix down. These tracks are: the dialogue track, the three FX tracks, the two ambiance tracks, two music tracks, and the voice over. We must do a pre-mix. We choose to do a pre-mix of the ATMOS and the FX-tracks. The Cue List looks like this:

START TIME	END TIME	VOL	SAMPLE NAME
23:59:57:20	23:59:57:21	+0	PIP
00:00:02:20	00:14:32:20	+0	ATMOS 1
00:00:02:20	00:14:32:20	+0	ATMOS 2
00:00:02:20	00:14:32:20	+0	MUSIC 1
00:00:02:20	00:14:32:20	+0	MUSIC 2

Figure 11. Pre-Mix of the ATMOS and FX-Tracks.

We call this pre-mix **MUSIC AND ATMOS pre-mix.**

Then we do the final mix.

The **Cue List** may look like this:

START TIME	END TIME	VOL	SAMPLE NAME
23:59:57:20	23:59:57:21	+0	PIP
00:00:02:20	00:14:32:20	+0	MUSIC AND ATMOS pre-mix.
00:00:02:20	00:14:32:20	+0	FX 1
00:00:02:20	00:14:32:20	+0	FX 2
00:00:02:20	00:14:32:20	+0	FX 3
00:00:02:20	00:14:32:20	+0	DIALOGUE
00:00:02:20	00:14:32:20	+0	VOICE-OVER

Figure 11. Final Mix.

If your computer is not be able to play all the tracks at the same time, you can work your way around this by loading the tracks into the **Transport** and reducing the **rate**. This enables you to do the mixdown while out of sync with the video, but when the final mix is done you just edit the **rate** back to normal, and it will play in perfect sync.

Scaling the Final Mix

Now it is time for the final check of the dynamic range. A traditional limiter needs time to react. When a loud sound is entering the limiter it will let some of it through almost unprocessed before it has had time to decide what level of limitation is needed. We call that the rise-time of the limiter. Because of that, we usually use far more limitation than we need just to compensate.

Using **scale** is a much better method. We examine the final mix in the edit window. If there are any peaks in the final mix it will affect the overall loudness of it. When the program is laid back to the video master, you must adjust the recording level in order to make headroom for highest level of the mix. If you use **scale** to adjust the highest peaks of your mix, the overall sound level will be higher and you will get a better signal-to-noise level on your video master. Your mix will sound louder.

Using scale as a limiter

This work can be time consuming, but it is absolutely worth while.

1. Locate a part of the mix that is too loud. Mark a **range** that is covering it and display it clicking **Show Range**.
2. Make a new range that starts where the graph shows that the level begins climbing and stops when it has reached the top.
3. Select **scale** and set the start to 100% and the end to how much of the sound level you want left after scaling. Try 80%. (That means that you are attenuating the sound by 20%.)
4. Click OK!
5. Grab the start of the **range** with the mouse pointer*, and drag it forwards past the endpoint of the range until the point is reached where the level has gone back down, thus making a new range.
6. Select **scale** again and this time set the start to 80% (the end level of the previous scale) and the end to 100%.
7. Click OK
8. Play it to hear if the adjustment is sufficient. If not, repeat the process.
9. Repeat this procedure for the parts of the mix that are too loud.
10. The highest part of your mix should <u>not</u> make the rightmost led on the digital peak meter light up. If it does, clipping may occur, and you should use scale to adjust the level of that part of the mix, or bring the overall level of the mix down.

* *If the* ***range*** *covers only a small part of the* ***edit window****, it may be difficult to grab the start of it with the mouse pointer. If you miss it Studio 16 assumes that you are making a new* ***range*** *and you will lose the previous endpoint! This can be disastrous when working with certain sounds and it is highly recommended to use* ***zoom*** *in order to make the* ***range*** *cover most of the* ***edit window****.*

3.7 AUDIO LAYBACK

Audio layback is the last part of audio post production. This is the last opportunity to control the sound. After spending maybe weeks of perfecting the sound, we can afford to spend a little time on this as well. Whether the audio layback is done directly from Studio 16 or from a master tape (DAT or other professional tape format), the levels must be right and it must be perfectly in sync.

In the professional world we use a reference tone of 1KHz for level. Different video formats have different specifications, and some television companies again have their *own* specifications. None of these will be dealt with here. The reference we will use is an internal reference within Studio 16. The only known level we need is the difference between the reference tone and the maximum program level.

Procedure for correct audio layback

We need a test tone of 1 KHz. A 1 KHz sine wave needs a sampling rate of 2000 Hz. There's no point in wasting hard-disk space.

1. Open the **Meters.**
2. Make sure that both the **analog** and **digital** is selected.
3. Open the **Recorder.**
4. Leave the mixer's **input** fader at 0 dB, and adjust the **Gain** until the VU-meter reads exactly 0 VU.
5. Select **Name** in the **Record Window** and call the sample **Test**
6. Record at least one minute.
7. Enter **Test** in the **Cue List** and select **Play**
8. Open the **Mixer**.
9. Check the level. It should now read 0 VU when both the **Play 1** and **Output** faders are in the 0 dB position.
10. Adjust the **output** fader until the two upper LED's of the **Meter Window** are lit.
11. Then adjust the **Output** fader back down until the upper LED goes out. Do this very slowly! Because of the peak hold function on the digital meters they need time to react.
12. Adjust the **Output** fader up and down until you are absolutely confident that you are at the exact point where one 1/4 dB (this is indicated on the fader) up makes the rightmost LED light up.
13. Read the dB indication of the fader. This, I have noticed, varies slightly from card to card, but should be about +4 dB. This reading is called **Maximum Program Level** or **Maximum Peak Level** and is referred to as **MPL**.
14. Readjust the **output** fader to 0 dB. If you are going on tape before you do the final audio layback, record the test tone for one minute in front of the final mix.
15. Consult the manual of the VTR to find the maximum input level.
16. Use the test tone from Studio 16 to find the correct input adjustment of the VTR.

The arithmetic is:

VTR Max input level
- MPL
Level of calibration on the VTR

Example. You have recorded the test tone at 0 VU on the VTR. The maximum input of the VTR is +9dB. The internal difference between the reference tone in Studio 16 and the maximum peak level (MPL) is +4dB. The arithmetic is:

  9dB **VTR Max input level**
- 4dB **MPL**

---

  5 dB **Level of calibration on the VTR**

That means that you adjust the VTRs input level to +5dB when playing the test tone from Studio 16. When you record the final mix from Studio 16 the maximum level will be +9dB on the VTR.

On digital recordings a reference tone for level is sufficient. Analog recordings should also have 1 minute of pink noise recorded for EQ adjustments.

3.8 SECURING THE SYSTEM / MAKING BACKUPS

About hard-drives

Using large hard-disks is considered to be convenient when working with samples, but with Studio16's ability to address as many as seven SCSI-devices, this is no longer so.
When using large hard disks it's advisable to make partitions that are no bigger than 500 megabytes. This makes the drive quicker and you destroy less if a breakdown occurs.
It is strongly recommended to have a program for repairing damaged hard disks. **Quarterback tools** has proved, on several occasions, to be excellent for repairing hard-disks.

Warranties

This is important. Your hard disks contain your work, and their integrity is crucial. Make sure to purchase your drives from a reliable dealer who can offer a replacement deal in case your drive should need warranty service or repair.

Power backup

There is always a risk of loosing data on a hard-disk. Being a professional means you can't afford to loose your work because of power failures. Have your computer connected to a battery backed up power-supply that allows you to operate your computer and hard-discs for at least 5 minutes after a power failure so that the system can be taken down without losing data.

Analog backup

Ideally backups should be made every 15 minutes or so, but since backups can take a while, and we don't want to keep our customers waiting, we normally don't do this. A good substitute for data backup is to do a quick analog backup. That can be done during a run through, and will not cause a delay.

Turn off the input monitor of Studio 16 when recording from the Cue List. It will add noise to your recording. Use a DAT player and record the output from the Cue List. That way you will save time if your hard-disk should crash or if by accident you erase your Cue List. There is no undo if you should, by accident, select save instead of load when wanting to load a Cue List.

The loss of sound quality is insignificant compared to loosing hours of work. I always backup from my Cue Lists using this method rather than using a data-backup program when I am working. Then I do a proper backup of the files that are important later.

If a hard-disk crash should occur while working I just sample the sounds needed as played from the Cue List, and make an entry to play it from the Cue List, synchronizing with the sync-pip. (And the customer never knew anything was wrong.)

DAT backup

DAT backup drives are an attractive archiving option. They are very fast, compared to other tape streamers, averaging about 12 megabytes per `. They are also very cost effective: The drive itself costing about $2000 and the DAT medium being very inexpensive. A DAT tape can hold about 1.3 GigaBytes of information.

Optical Drives

Probably the best storage medium is the optical drive. An optical drive connects to your system like any SCSI-device and is capable of storing 1 GigaByte of information on removable high-capacity optical cartridges, (500 megabytes on each side). Although you cannot use optical drives for direct to disk recording with version 2.0 of Studio 16 you can play two channels of sound direct from an optical drive provided you have enough RAM to set the buffers for 1024K. Optical drives are by far the most flexible, but they are still rather expensive.

3.9 USEFUL PERIPHERALS

These are all products I have used and tested over a period of time and found to be well suited.

The Fostex 4010 / 4011
The combination that will cover all aspects of timecode handling.
The 4010:
Timecode generator. Generates timecode from an internal crystal, external pulse, or a composite video signal.
Time code regeneration.
High speed timecode reader.
Generates CTL pulse from timecode or external sync.
Generates and reads all timecode formats.
The 4011:
Inserts timecode and userbits from LTC and VITC.
Generates VITC from LTC.
Requires the 4010 for operation

Brainstorm Electronics Inc. SR-1
Timecode refresher.
Refreshes timecode in any format with adjustable curves for SMPTE, EBU and Square Wave.
Does not regenerate!
Balanced input / output and adjustable gain.

JL Cooper PPS 100
SMPTE to MTC converter and event generator.
Generates timecode in all formats
Generates sync pulse with adjustable PPQN frequencies.
Generates event pulses for trigging CD players and digital delays, etc.

Tascam MTS 1000 MIDiiZER.
Synchronizes tape recorders to video or any other TC source.
Is a complete remote control for Tascam multi-track recorders.

Generates timecode in all formats.
Generates MTC from all SMPTE/EBU formats.
Controls tempo and synchronizes midi sequencers to master or slave.
Reads timecode backwards and forwards. High speed reader.

Since monitoring, from my point of view, is the most important part of any audio production setup, I will recommend the combination I am presently using for close monitoring

Klipsch KG 1

A very nice speaker that has a very accurate reproduction on all frequencies and has a clear and distinct bass.

Electrocompaniet 100

Beautiful amplifier that works very well in combination with Klipsch KM 1.
This combination is excellent for monitoring audio for TV programs and commercials.

4. About the Author

Øistein Boassen started working with film & video as a freelancer in 1974. Since 1980 he has designed and built two studios for audio post production and has been head of the sound department of a major Norwegian film company. Since 1989 he has been teaching the use of audio and music at universities and film-schools in Norway. He has done sound editing and final mix on several motion pictures and has composed and performed music for several films as well as four full-length ballets.

In March 1992 he started an audio post production company in Oslo, Norway, called The Sound Factory. To date, The Sound Factory has built two digital sound editing rooms, one advanced MIDI-suite and one mixing theater and screening room where an Amiga 3000 with Studio 16 are the main editing tools.
The Sound Factory works with 35/16 mm film as well as video. The Sound Factory is doing dialogue replacement, effects-dubbing, dialogue sweetening, music composing and editing, final sound mixing and audio layback to all professional video formats to customers from all over Scandinavia.

The Sound Factory will be happy to answer any questions you may have regarding the use of Studio 16 in audio post production.

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Appendix E

Pro-consumer Audio-for-Video Applications

We asked a local videographer and user of Studio 16 to include some of his experiences on incorporating Studio 16 into his video suite. Professional and consumer videographers will be interested in the following suggestions and experiences of Rob Grant from Video InfoCom in San Jose.

This document was originally written by Rob Grant when using Studio 16 1.0 and an AD1012. The text was later edited by SunRize to reflect changes in release version 2.0 of Studio 16 and the availability of the AD516.

INTRODUCTION

In the video world, quality audio is often ignored. The reasons for this are many, but the primary reasons are those of expense and technical complexity. Studio 16 goes a long way in the resolution of these difficulties and, with its proper use, quality audio and audio effects can be economically incorporated into your video productions.

Audio-for-Video Hollywood Style vs. Most Professionals

Of course, the ideal situation is to do things Hollywood style. In Hollywood, most television producers worry about the video components first and the audio second. Given Hollywood budgets, they can afford to do this. Typically, after a television sequence is filmed, the video is sent to a musical director/composer who quickly (usually overnight) puts together a custom musical score which progresses in theme and synchronization with the video. The television show producer, the musical director/composer, sound engineers, and musicians then gather at a local sound studio and via an arduous and iterative process, lay the audio onto the video tracks. For about $40,000/day you could do the same.

Obviously, the expense of this technique is beyond the means of almost everyone. However, the kinds of alternatives I will discuss give excellent results and at a price most Hollywood producers would be embarrassed to consider.

As mentioned above, the Hollywood technique usually involves the creation and editing of sound to meet the requirements of the video footage. The practices demonstrated in this chapter involve not only these techniques, but also and perhaps more importantly, editing and composing video segments to meet the requirements of existing audio tracks. You may wonder how Studio 16 helps with video editing, but, as you shall see, Studio 16 allows us to determine exactly where certain sound characteristics are located in the audio track and thus how long our video segments should be.

As the title of this chapter indicates, we are talking about the use of the Studio 16 product for pro-consumer purposes and therefore we will assume that the user has access to video equipment with SMPTE time code capability. However, many of the SMPTE capabilities that are required for SMPTE use are included in Studio 16. Therefore, if you lack SMPTE equipment other than the AD1012 or AD516 board, but have the patience for some trial and error, you can still use many, if not all, the techniques I am going to describe. Having said this, I will, nevertheless, assume that the user possesses full SMPTE capabilities and accurate, repeatable VTRs (video tape recorders).

Typical Audio/Video Problems

For most purposes, the task of combining audio and video poses a common set of problems. These problems usually involve the following:

- Music Characteristic Determination - deducing the tempo of a music selection and the location of particular audio characteristics such as the crashing of a cymbal etc.

- Synchronization - starting and stopping audio tracks precisely at a given video frame, fading the audio from one level to another within a specified number of video frames, etc.

- Mixing - controlling the percentage contribution of any separate audio track to the total volume level at any given moment.

Studio 16 Capabilities and its Common Use

Studio 16 contains modules which allow you to effectively solve all of these problems and many more as well. Assuming that you are already familiar with the basic capabilities of the Studio 16, let me briefly revisit some of these tools to discuss a few of the capabilities that we will use in the examples which are presented latter in this chapter.

Recorder
The Recorder is used to record the original sounds; usually from CDs, microphones, and/or MIDI controlled music equipment. The sampling rate for the recording should be chosen to match the fidelity of the playback equipment and thus avoid wasting valuable disk space while insuring acceptable audio quality. On my AD1012, I use a 32,051Hz Sampling Rate and a 16,667Hz Filter.

Meters
The meters are used to verify that the record and playback levels of each audio track are in correct proportion (mix) to each other and free from clipping.

Editor
In these examples, the Editor is used primarily to cut, copy, and paste audio tracks, eliminate leading and trailing blank space, and control fading. Importantly, the Editor, in conjunction with the SMPTE Monitor and SMPTE time code capture features allows specific points in the audio track to be located and synchronized precisely with the video track. In addition, these same features allow various audio characteristics and timings to be determined so that either the audio or video tracks can be recorded and/or edited to achieve the producer's desired effect.

SMPTE Monitor and SMPTE Generator
These modules are used to verify that the editing process has been performed correctly and in a manner which achieves the desired effect while avoiding the hassle, tape wear, and delay of running the VTRs, or other SMPTE time code generating and reading equipment. The SMPTE Generator is particularly valuable in that it can trigger all of the Studio 16 SMPTE related modules. Specifically, this means that sounds on independent tracks can be initiated, mixed, faded, etc., in precise relationship to sounds on other tracks.

Transport and Mixer
The Transport controller has many functions; however, there are two functions that you will probably find most important for use in video work. The first is the ability to assign given sounds to any one of the playback tracks. The second is the ability to playback up multiple tracks while simultaneously recording the combination of the tracks to another track. This second capability is particularly important when your final sound is to be composed of sounds which originally came from more many sources. Since AD1012 can only playback four tracks (maybe less if you have a slow system), you may need to combine tracks in order to playback all the sound components you desire. The contribution of each track to the total sound volume recorded on the fourth track is controlled by the Mixer volume setting for each track.

Cue List
This module is used to playback sounds in synchronization with SMPTE time code and thus with your video track. The playback of up to eight independent sound tracks may occur simultaneously with the AD516, (4 tracks with the AD1012). If one of the sound tracks ends during playback, the channel (or track) it occupied becomes available for use by another audio track. Thus, the Cue List may contain more than four entries as long as no more than four sound tracks are active simultaneously. Again, if you need sound from more than eight/four sources simultaneously, you would use the Transport and Mixer to combine two or more of these sounds into a single track. Of course, since Studio 16 is digitally based, you suffer little, if any, impact on sound quality during the mix down process and, therefore, the price of performing a mix down operation is reflected only by the amount of time it takes to complete the

operation. Most often though, you will find that eight track playback is more than adequate for commercial tasks and, therefore, the mix down task is unnecessary.

CONFIGURATIONS

Figures 1 and 2 show inter-connection diagrams for using the Studio 16 product with Longitudinal Time Code (LTC) and Vertical Interval Time Code (VITC) respectively. Of course, if your equipment can utilize VITC then you should typically do so as it has several important video editing advantages over LTC. Among these advantages are its ability to accurately display a frame's time code over a broader range of editing speeds (particularly while a tape is paused) relative to LTC. The only notable disadvantage of VITC use with the AD1012 is the requirement of a VITC to LTC translator. These translators can be obtained at reasonable prices. (I use the Horita VLT 50 - $289 at the time of this writing.) If you are editing with VITC and do not have a VITC to LTC translator, nor wish to purchase one, then you will have to stripe your video tape with LTC; a straight forward but time consuming task.

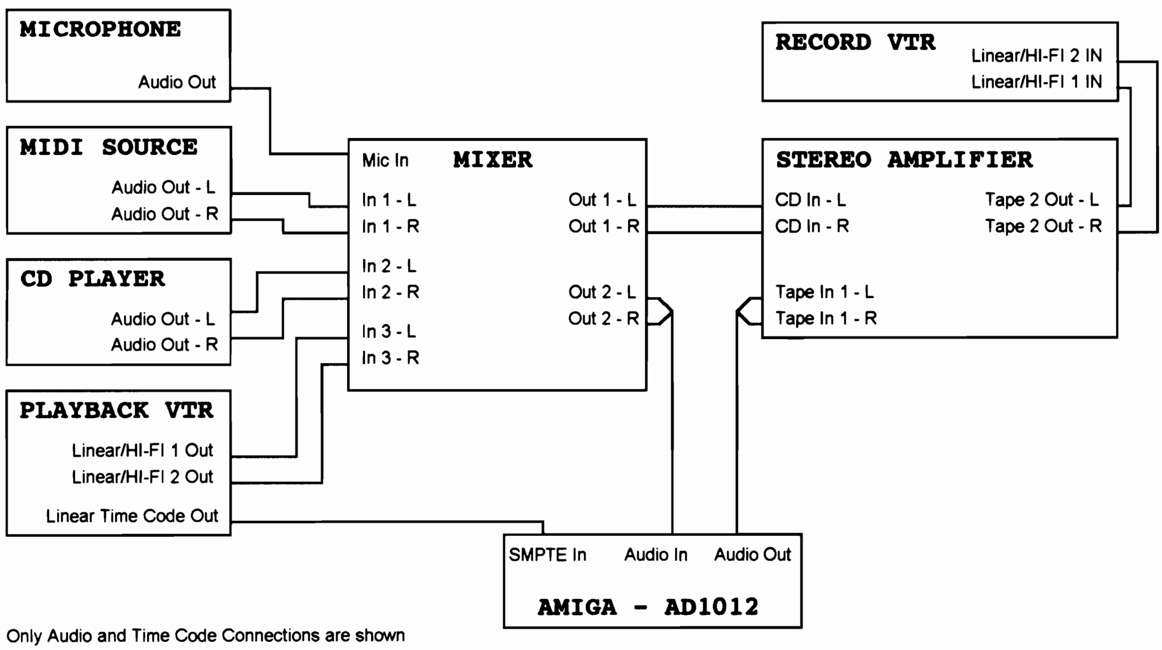

Figure 1. Typical Audio Setup for LTC

LTC

If your editing suite is set up for use with LTC, then you can connect your LTC output directly to the AD1012 or AD516's SMPTE input as shown in figure 1. In reality, you would most likely have the LTC output connected from the Playback deck to a distribution box or into other decks before feeding it to the AD516/AD1012. However, if you are connecting your equipment solely for the purpose of putting audio onto your master video tape then the configuration shown is appropriate. If you are using VITC and a VITC to LTC translator, inter-connect your equipment as shown in figure 2. If you are using VITC for editing and do not have a VITC to LTC translator, then inter-connect your equipment according to figure 1 and stripe your video tape with LTC time code.

Figures 1 and 2 reflect configurations which suit our needs. Obviously, there are many configurations and equipment selections that can be used and you will most likely interconnect your components in a manner which is appropriate to the applications you deal with most frequently. In addition, keep in mind that the figures show only the interconnections of the audio and time code signals. Video connections vary widely and should be dictated by the specifications in your video equipment manuals etc. Both diagrams utilize a mixer and stereo amplifier. Again, the diagrams illustrate configurations which match the features and capabilities of our equipment. You may either desire or be required to alter your configuration based on the capabilities of your equipment. Particularly, you may want to feed the audio output of the AD516 or AD1012 back into your mixer so that you can mix the output with signals from any of your other signal sources. If your mixer can be connected in this manner while maintaining isolation between the input and output of the AD516/AD1012, it will prove useful.

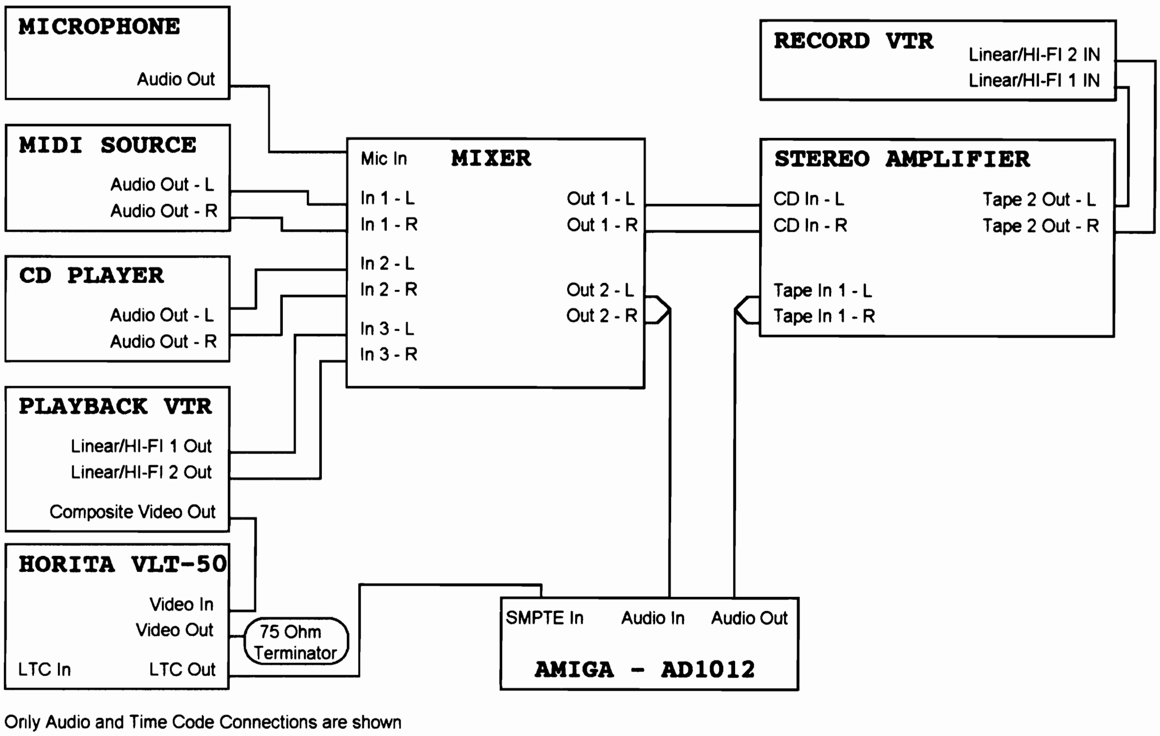

Figure 2. Typical Audio Setup for VITC

Unfortunately, the simple mixer we used would not work properly when connected this way. In addition, our stereo amplifier - a Yamaha R9 receiver - provides the ability to feed input to any one of two tape outputs. This feature was utilized in our interconnection schema. Obviously, if you do not have this capability you will have to make some adjustments.

Professional specifications almost always dictate the requirement to record sound on a video tapes HI-FI tracks. Yet, as there are still some consumer decks in use which do not have HI-FI playback capability, it is often necessary to record sound not only on the HI-FI tracks, but also on the lower fidelity linear tracks. The decks we utilize have inputs which will record sound to both the linear and HI-FI tracks simultaneously, and the figures reflect this type of inter-connection - you may, once again, need to alter these connections to suit your particular needs and equipment capabilities.

VITC

An examination of figures 1 and 2 reveals that they are essentially identical with the exception of the VITC to LTC translator and the time code interconnections. In our setup, we utilized the Horita VLT-50 VITC-LTC translator. The use of a VITC to LTC translator is straight forward and adds little complexity to the overall configuration. However, two particular points are worth mentioning. The translator reads in a composite video signal which, as the VITC name implies, has the time code information embedded in the Vertical Blanking Interval. For this reason, the time code signal should be routed to the translator before passing it through any equipment such as a Time Base Corrector (TBC) whose function often strips VITC time code information from the signal. Some VTRs (the Panasonic decks we used, for instance) have internal time base correctors whose use requires that the composite video signal pass through the TBC before the signal is output to the external environment. In our experience thus far, such VTRs appear to have been designed so as to leave the time code information intact and, therefore, do not adversely affect the translators performance.

The second point worth mentioning concerns video signal termination. Reliable operation of video equipment almost always requires that video signals are properly terminated with a 75 Ω resistance. This practice is certainly a requirement with the equipment and configuration shown in figure 2. In figure 2, the composite video signal from the VTR is fed into the VLT-50 video in connector. Although the VLT-50 has a video feed through output, it was not utilized in our configuration. As the figure shows, reliable operation of the VLT-50 requires that the video output be terminated. Otherwise, the signal level received by the translator exceeds the translators design limit for reliable operation and the AD516/AD1012 will not receive a consistent LTC signal. One indication of a poor LTC signal at the SMPTE input is immediately manifested by a blinking pixel in the upper left hand corner of the SMPTE Monitor in the Studio 16 software.

EXAMPLES

Important Comments

Most computer software users (myself included) have the innate instinct to dive right into a program and begin experimenting with its features before reading the appropriate documentation. Such an approach is often more productive than the laborious alternative of documentation digestion. Those who use this procedure with the Studio 16 product will meet with a significant amount of success. However, when you wish to become truly conversant with all the Studio 16 advanced features you will most often find it quite helpful to read the manual and try the examples. This is true not only from the perspective of gaining insight into the products full capabilities, but perhaps even more importantly, becoming familiar with the operational peculiarities which all products exhibit as a byproduct of their creator's "Stream of Consciousness" and other technical dictates. Particularly, the Amiga file system does impose some restrictions on the speed of data acquisition and the acquisition methodology.

In the computer software based world, data can be stored primarily in two areas - RAM or Disk. While RAM offers significant speed advantages, expense and hardware limitations often render RAM based manipulation and data storage alternatives for audio applications unattractive. This is easy to understand when you consider that a single track of high fidelity sound (un-compressed) requires about five megabytes of storage space per minute of recorded sound. For this reason, the Studio 16/ product offers the advantage of being primarily a disk based system. However, while this fact is very beneficial, it does impose a few operating characteristics that will quickly become apparent in the audio-for-video editing process and which you should be cognizant.

Those who are accustomed to video/audio editing from a VTR source expect sound to begin immediately upon hitting the play button. With Studio 16 this behavior is often not exhibited. The reason is simple. Studio 16 must load sound from disk and exhibits a starting delay which is proportional to the location of the desired start point from the beginning of the audio track and your disk speed. Thus, the further the desired start point is from the beginning of the audio track, the longer it will take to begin playback. Therefore, when trying to locate a sound feature or make an edit you must begin playback of the video tape in advance of the desired edit point. Exactly how far in advance the video must be started depends on the length of the audio track and the location of the edit point. The longer the audio track is and the farther the edit start point is from the beginning of the audio track, the larger the required advance time will be. Typical advance times for tracks which are three minutes in length and with desired edit points in the last thirty seconds of the audio track are in the range of 10 to 20 seconds. On the other hand, edits on these same audio tracks which occur at the beginning of the track require advance times of only three or four seconds. This delay is a byproduct of disk based systems and although sometimes inconvenient, allows the Studio 16 product to edit audio tracks of a length far greater than those which can be handled by similar RAM based systems. In fact, the massive data storage requirements of multi-track editing and/or playback make the use of a disk based system a necessity.

Like most LTC based editing products, the AD516 and AD1012 hardware can accurately read LTC time code only when the time code is running at normal play speed (this applies to the VITC-LTC translator as well). When you use a VTR's shuttle or jog controls, the AD516/AD1012 will temporarily loose track of the video tapes SMPTE time code location. However, once you begin playing the tape at normal speed, the Studio 16 will re-sync to the time code.

General Guidelines

The examples which follow are based on the actual work that Video InfoCom - a San Jose, California video production company - performed as part of the preparation of their company introduction and demo tape. Video InfoCom used the equipment configuration illustrated in figure 2. In the examples which follow, a presentation of the scenario and desired effect will be described followed by the steps taken to produce the final result. Royalty free CD music and sound effects were used heavily with the occasional accompaniment of various sounds generated by a MIDI controlled synthesizer. All the sounds discussed in the following examples were mastered on the linear and HI-FI tracks of a Panasonic 7750 S-VHS VTR. As the specifications for this deck indicate a flat audio frequency response to 15 KHz and given that the average ear does not hear sounds above 16 KHz, a sampling rate of the AD1012 was set to 32,051 samples per second and a filter cut off point of 16,667 Hz were selected. These choices allow disk space to be conserved while assuring the reproduction of quality sound.

A few final remarks should be made regarding these examples. There are many ways to obtain the desired effect listed in each example scenario. Obviously, not all the possibilities can be covered. Therefore, do not be surprised if you discover a faster, more efficient, or just different, way to accomplish these tasks. The best way to attack and execute any problem varies widely with specific circumstances. Nevertheless, there are certain basic techniques and approaches to solving various audio-for-video problems and those techniques are illustrated in the following examples.

Example 1 - Editing and Synchronizing Short Sounds to Repetitious Video

SCENARIO

As part of the animated opening sequence for a demo video, alphabetic characters spelling out the name "Video InfoCom" spin off the surface of a rotating planet and tumble out towards the viewer eventually impacting the face of the viewing screen. Following the impact of the last character with the screen, the entire animation - spinning planet and all - gradually fade to black. In order to complete the production master tape, appropriate audio tracks must be added to the pre-recorded video. The film's producer has asked you to accomplish the following tasks: First, he would like a constant ominous tone which starts five seconds before the video segment and continues until five seconds after the video segment completion. Second, he would like nighttime jungle sounds to begin at the start of the video segment and end abruptly in sync with the last character impact on the screen. Third, as each alphabetic character hits the viewing screen, the producer wants a cannon blast to sound.

The completion of these tasks will be fairly easy with Studio 16. Only the editing of the cannon blasts so that they occur in sync with the impact of the alphabetic characters will require any extra effort. The common approach to completing this assignment is to record, mix, edit, and synchronize all the sound on hard disk first and then record the audio from hard disk to the VTR as the final step.

STEP 1 - Locate or Create your Basic Sounds

The first task that must be accomplished in order to complete this assignment is finding and recording the desired sounds. The sounds for this example were obtained from three sources. The ominous, low, background tone was generated by a synthesizer and recorded by the Studio 16 software under the track name *PF13*. As mentioned above, the AD1012 sampling rate for recording was 32,051 samples per second and the filter was set to 16,667. Obviously, you must know how long to record a sound. To determine this simply playback the video on the VTR and note the SMPTE time code starting and ending points. At this point, only the approximate duration is required.

Since the producers original specification asked for the low, ominous tone to begin five seconds before the video and end five seconds after the video fade to black, we must add ten seconds to the previously noted video length in order to determine the record duration. Thus, if the video lasts 18 seconds - as was actually the case, we would need to record the *PF13* sound for a total of 28 seconds. In reality, the sound should be recorded for a longer period because, as we shall see, it will need to be edited later. You should always record the sounds longer than necessary because there will almost always be errors in the start and stop times on the recording which will have to be edited out.

Once the ominous, low tone (*PF13*) is recorded, the other sounds must also be captured. The second sound source was a track from an animal and environment sound effects CD of nighttime in the jungle. The track was recorded under the name *Jungle*. The third and final source of sound, the cannon blast, was also from a sound effects CD and was recorded to disk under the name *Cannon*. In actuality, two cannon blasts were on the cannon CD track and both were recorded. Latter, the desired portion of the track was isolated in the Editor.

STEP 2 - Editing the Samples

One of the most common and important uses of the Editor is simply to eliminate leading and trailing blank space from an audio track. In order to insure that none of a sound is lost during recording, it is necessary to start the Studio 16 record mode before an audio source begins playback and to end the Studio 16 record mode after an audio source ends playback. Such is the case not only with Studio 16, but with any recording device. As a result, there will almost always be a leading silence and ending silence on the newly recorded audio track. Since you want a sound to begin immediately when triggered, and since you want to conserve disk space and free up tracks as soon as possible, it is important to eliminate the leading and trailing silence.

This task is accomplished in the Editor. After an audio track is loaded in the Editor, look for the leading silence period, then alternate between the **Mark Range**, **Show Range**, and **NonDestructive Cut** functions until you have eliminated the silence. Follow the same procedure for trailing silence.

In this example, the producer's specification for the *PF13* and *Jungle* tracks is straight-forward; it requires only that the length of each track be cut to meet the specified requirement. Therefore, by loading each track, eliminating leading space, and cutting the duration of the audio to meet the length determined by the specification, the edit tasks for the two tracks are completed. There is no need to eliminate the trailing space because this space will be cut out when the entire audio track is cut to length.

A far more interesting set of edits is performed as part of the series of steps required to produce the repetitious cannon blast audio track. An examination of the video track shows that each letter of the word "Video" flies up separately and impacts the viewing screen followed by the entire word "InfoCom." A further examination of the video tape shows that each letter of "Video" and then the entire word "InfoCom" hits the screen four video frames apart. Since we are using Non-drop frame SMPTE time code, this means that the four frame spacing corresponds to 4/30 or .1333~ seconds between object-screen impact. Therefore, our audio track must be edited so that there are six consecutive cannon blasts with each one following the other by .1333~ seconds.

Calling up the Studio 16 edit screen with the original cannon sound track selected we see that of the two cannon blasts shown, cannon blast one seems to have a more abrupt start and is shorter in duration. Therefore, we will select it as the source of our sound. In order to isolate this blast so that we can get a clearer view of what we need to do, the first step is to cut all of the sound following this blast from the track. This is done simply by marking the range we wish to cut and selecting **NonDestructive Cut**. In addition, we will eliminate the leading space from the track by marking a leading range, cutting, zooming in with **Show range**, marking and cutting again, etc., until we have eliminated all the leading space accurately. It is important to go through this process iteratively, so as not to cut any of the actual desired sound. Usually, three or four iterations is enough.

Since we are working with video, it makes sense to perform all editing using SMPTE time code instead of samples per second. This practice will save a lot of tedious conversion. In order to have all points, ranges, and lengths, displayed in SMPTE time code, deactivate **Units in Samples** from the Editor's Option Menu. Along the top of the edit screen are a series of numbers which include the start and end of the display and the start and end of the currently marked range.

We can see that the length is 2 seconds and 10 frames. According to the calculations we made earlier, we can use only 4 frames of the cannon blast for each screen impact; therefore, only a very small portion of the waveform will be utilized. Use the grid as a guide to mark a 4 frame range. Clicking the **Play Range** button plays the marked range and reveals that this short marked range sounds little like the original complete cannon blast. Nevertheless, this is as much of the sound that can be use within the producer's guidelines and, as will be seen, when six of the marked regions are placed back to back as the video requires, the resulting machine gun sound fulfills the producer's requirements. It is important to remember that listening to <u>short ranges</u> of sound can easily mislead you when trying to determine what a complete audio track will sound like. More often than not, the completed audio track will produce acceptable results; therefore, in these cases, rely more on your specifications and measurements than on your ear.

At this point, we need to make a copy of the 4 frame range so we can add five copies to the existing marked range. This is done by selecting the **NonDestructive Copy** option. Once this is completed, we mark a new range which starts with frame 4 and extends through range 7, thus giving us another 4 frame range (frames 4, 5, 6, and 7). To insure that you find the beginning and/or end of any given frame, you should activate the Grid to display on frame increments from the Editor's Option Menu. You may have to zoom in on the graph to see the grid.

After marking the destination range, select **NonDestructive Paste - Replace**. Repeating the paste operation a total of five times renders six cannon blasts (the original plus five copies). The six cannon blasts occupy frames 0 - 23 (24 total frames). The waveform looks quite repetitive through the first 23 frames. Occasionally, a feature appeared in figure 6 which did not seem to repeat. This is because the graph is just a rough visual guide. The actual sound is duplicated exactly. The copy and paste procedure used in this example is graphically based. A more accurate method of performing the copy and paste involves using **Set Range**. Selecting **Set Range** from the Options Menu allows the precise designation of the start and endpoints of a range by entering numbers in the appropriate boxes that appear when the button is pressed.

Returning to our example, the audio track could be cut at the start of frame 24 and this audio track considered finished. However, the portion of the audio track from frame 00:24 to 02:09 provides a nice fade that goes well with the video. Therefore, a decision is made not to cut the track but to leave it as shown in figure 6. In order to save this edited waveform, the entire wave is ranged. **Make Permanent** was selected to create a final version without "non-destructive" edits. This, however, is not required.

STEP 3 - Audio Track Playback and Verification

In order to complete this audio assignment, all that is needed is that each of the sounds be set-up in the Cue List such that they execute at the appropriate times. An externally supplied SMPTE time code signal (generated by the playback VTR) will be used to trigger each sound and the resulting sound will be sent to the record VTR where it will become the final sound track of the Production Master Tape.

The first step in this final phase is to call up the Cue List by selecting it from the Applications Menu. Next, from the Open List, we drag the name of the first track we wish to playback. Then enter the SMPTE time code at which the track should initiate, and the volume at which we want the track to playback in the edit fields at the bottom of the Cue List. The procedure for entering the rest of the tracks to be played is similar

Once all three sound tracks (*PF13, Jungle, Cannon*) are entered into the Cue List with their corresponding volumes (in this case, all set to 0 dBs) and start times, you will want to verify that the resulting sound is correct before recording to the Master Production Tape. This can be done by supplying an external SMPTE time code to the AD516 or AD1012 board or by using the internal SMPTE Generator. For verification purposes, it is almost always more efficient and convenient to use the SMPTE Generator. In order to use the SMPTE Generator it must be selected from the Applications Menu. I also find it useful to display the SMPTE Monitor. Whenever the SMPTE Generator is open, it becomes the active source.

Once these actions are completed, all is ready to begin playback via the SMPTE Generator trigger. If the time code at which your first sound is to trigger is far from zero (such as a start point of 11:30:00) you will want to move the SMPTE Generator counter to this point rather than wait for the generator to methodically count to the start point. This is done by repeatedly clicking on the SMPTE Generator fast forward button. Each time you click on this button, the count speed is doubled. Therefore, repeated clicks on the fast forward symbol will rapidly move the counter to the desired point, at which time you may stop the count by hitting the stop symbol (you may need to rewind or modify the counter a little once you have stopped it in order to get the counter precisely to the point you desire). You should <u>always</u> set the counter a minimum of 2 seconds ahead of your first audio tracks start point so as to give Studio 16 enough time to load the sample. This implies that an audio track can not begin at a SMPTE time code prior to 2 seconds. Although this requirement may seem restrictive, in actuality it is not, as all professional videos are recorded with at least 20 seconds of lead time before the start of the video track.

Once the SMPTE Generator is set according to these guidelines and the Cue List is turned ON, you may begin playback of the sounds by clicking on the Generator's **Play**. Via this process, you may easily verify the performance of your audio tracks before committing the sound to tape. Any problems in synchronization can be corrected by adjusting the SMPTE start points of the appropriate track or tracks in the Cue List. Problems with the actual sound characteristics are addressed by a return to the Editor where the required changes can be made.

After audio performance is verified via the SMPTE Generator, it is time to check its synchronization with the recorded video. Close the SMPTE Generator, and connect the your playback VTR to the AD516 or AD1012 SMPTE in, the audio track will now be triggered by and synchronized with the playback deck. Via this operation, the performance of the audio/video tracks can be evaluated.

I occasionally notice that the audio tracks did not start at the appropriate time code set in the Cue List. Most often, it was the result of multi-tasking conflicts between multiple programs running on the Amiga concurrently. Usually, these problems were resolved by shutting down unnecessary functions in the Studio 16 software that hog CPU time (i.e., Meters, SMPTE Monitor) and/or closing other programs that are running on the Amiga. Sometimes it was necessary to re-boot the Amiga.

STEP 4 - Committing Your Audio Tracks to Tape

After the performance of your audio track has been verified, it is time to record your audio to your Master Production Video Tape. This can be done by the same procedure used above to verify audio/video performance with the exception that now your record VTR is actually set to record both audio and video. However, you may desire to combine the separate audio tracks (*PF13*, *Jungle*, and Cannon) into one master sound track. In this example, the mixing is not necessary; however, the Transport module can be used to mix the Cue List down to one or two tracks. (Two tracks are only available for the AD516.)

Example 2 - Video Editing for Synchronization with Existing Audio

SCENARIO

Your producer has located a music selection that he would like to use in the introduction of a business video. The producer has stipulated that he would like a series of fast video cuts from scene to scene that occur in time to the music. On occasion, he would like to use a special wipe or video transition other than a cut and usually, it turns out, this special transition is not coincident with the rhythm of the music but rather on some other feature of the composition such as the sounding of chimes.

After examining the producer's requirements, you decide that the primary task you must perform is the determination of the timing of the rhythm and the location of the special composition features. Of course, since you are editing the video tape with SMPTE time code, you must define these characteristics in terms of frame numbers. In addition, you must decide which special transitions you will use - a decision that will be largely based on the determination of which of the available transitions will fit in with the theme and timing of the composition. Further, in the exercise of your right to a certain amount of artistic license, you have decided that transitions should not occur on every beat of the music, but should occasionally be separated by a longer period of time in order to promote visual interest; the producer agrees.

Recording and Locating Music Characteristics

The producer's musical selection is a song entitled "Logjam" which is distributed as part of a royalty free CD library. The first step you complete is to record this selection under the name *Logjam* using either Transport or Recorder. Since the use of Recorder is the simplest way to complete this task you utilize its capabilities to record this song in a manner which has been described in previous sections of this manual. Next, you load the *Logjam* recording into the Editor and eliminate leading and trailing blank space.

When a sound track is loaded into the Cue List, it is assigned a start time of 00:00:02:00. Most often, you will want to re-reference the audio tracks start time to the start time of the video. For instance, if your video segment started at 11:30:00, you would want to set your audio track to begin at this point as well. This not only allows the audio to playback in sync with the video when the audio track in the Cue List is triggered by VTR SMPTE input, but also allows any edits in the Editor to display the location of the edit or sound characteristic with reference to its location in the video. The Editor provides a mechanism which allows it to reference points in its audio tracks directly where they will occur on the video tape. This mechanism is simply the input of an offset and is entered by selecting the **Set Display Offset** in the Options Menu. Thus, in this case, you would enter 11:30:00 for the offset as this is the frame number on the video tape where this song is to start playing. Once this is done, you will notice that all location readings in the Editor are now referenced to this offset of 11:30:00.

The quickest way to locate a particular sound within an audio track in time code is to utilize a simple but powerful feature of the Studio 16 software called **SMPTE Capture.** The use of the **SMPTE Capture** feature requires that the SMPTE Monitor, SMPTE Generator, Cue List, and Editor modules all be displayed.

Once the various modules are set up, the use of the **SMPTE Capture** feature is fairly straight forward with one or two exceptions which will be detailed below. First, you set the SMPTE Generator to a time several seconds before the start of the song characteristic you are interested in locating. In this case, since the song begins at 11:30:00, we'll set the SMPTE Generator to 11:22:00. Next, click **Play** on the generator and **immediately** thereafter click on the SMPTE Monitor window to make it the active window. The SMPTE Monitor must be the active window (signified by a highlighted Title bar) in order for the **SMPTE Capture** feature to work. At this point, both the SMPTE Monitor and the SMPTE Generator should be counting but the music should not yet be playing (this is why the SMPTE Generator is set several seconds before the music characteristic of interest occurs - this provides enough time to perform the above actions prior to the occurrence of the music characteristic). Shortly the music will start, and when you hear the characteristic you are interested in you can accurately capture the SMPTE time of its occurrence by selecting Freeze Display from the SMPTE Monitor Menu or by typing Right Amiga-F. The time of the characteristic's occurrence will be captured in the SMPTE Monitor display.

The utilization of the **SMPTE Capture** feature in this way can usually capture the occurrence of an event with acceptable accuracy; however, since there is some error based on individual reaction time, you may want to further increase accuracy by marking a range in the Editor around the number displayed in the SMPTE Monitor and expanding it with the Editor's **Show Range**. Then, via the utilization of **Play Range**, you can further increase accuracy by visually repeating the playback of the range until you have a solid idea of the characteristic's location in time code. By using the **SMPTE Capture** feature, you can locate any music characteristic relative to the video's actual SMPTE time code and ,thus, video segments or transitions can be accurately edited to fit the requirements dictated by these audio characteristics. Also note the you can capture a specific time frame in the Cue List by using the TC Add Option.

The **SMPTE Capture** feature can be easily used to determine the basic tempo of a song. Once we have set-up the various modules as described above, we start the SMPTE Generator by clicking the **Play** symbol. Once the music starts, we count out 20 or more beats and then hit the space bar. Performing this operation using 20 beats with the *Logjam* sample will result in the SMPTE Monitor reading 11:42:06. Subtracting the start time of the song (11:30:00) from 11:42:06 leaves us with the duration of the 20 beat segment of music or 12:06. Since we are using Non-drop frame SMPTE code this further tells us that each beat takes $((12 \times 30) + 6) \div 20 = 18.3$ frames or .601 seconds. Therefore, in order to meet our producer's requirement to perform cuts on music beats you would perform the cut about every 18 frames. Because of the fact that the beats actually occur every 18.3 frames you would probably perform the third cut after 19 frames. Following this procedure would still leave us with a slight and probably unnoticeable error (.1 frames every 3 cuts). But, if you wanted to play it safe you would occasionally (most likely after every 10 edits) isolate a beat precisely using the **SMPTE Capture**, **Show Range**, and **Play Range** and make your next video edit to accurately re-synchronize your video and audio tracks. At Video InfoCom, we have used this procedure quite successfully.

The above procedures allow the determination of the tempo of a song as well as the isolation of any specific song characteristic in terms of SMPTE time code. Using these techniques, you can cut and paste video segments to match any audio track. As a result, you are now in a position to fulfill all of your producer's requirements.

Example 3 - The Cross Fade

SCENARIO

Your producer has given you a tape that contains two video segments with two different themes. The video segments transition into each other via a 10 second dissolve. The producer has chosen to accompany video segment #1 with the *Logjam* audio track, and video segment #2 with a sound track called "49er". The producer wants you to lay these two sound tracks onto the tape such that *Logjam* fades down from full volume to zero volume starting at 30 seconds and finishing 10 seconds latter and the *49er* sound track goes from zero to full volume starting at 30 seconds and also lasting 10 seconds. The completion of this task will allow the audio fades to occur in sync with the video dissolve.

This requirement will be easily fulfilled via the Editor's **Scale** and Cue List. In order to fulfill the requirements of this assignment, you decide to playback each of the songs on separate tracks by assigning them to the Cue List and triggering them at the appropriate SMPTE time codes. *Logjam* will begin at SMPTE time code 00:00:02:00 and *49er* will begin at 00:00:32:00.

Editing

The first step that you must complete is to record the Logjam and the *49er* soundtracks as described earlier. Next each sound is loaded into the Editor and the leading and trailing blank space is eliminated. Following this, you again load the *Logjam* segment into the Editor and cut it so that it is 40 seconds or 1,200 frames long. Now you must create a fade in volume. To do this, you mark a range on the waveform which begins at 30 seconds and ends at the finish of the sound track (40 seconds). Then you select **Scale** in the Editor and enter a starting volume of 100 and a finishing volume of 0. Upon hitting **OK**, Studio 16 will calculate and make the necessary changes to the waveform.

This process is repeated for the *49er* soundtrack except that the first 10 seconds of the *49er* soundtrack are edited via the **Scale** function. The input to the **Scale** function is such that it begins at zero volume and finishes at 100. Since the producer did not specify a duration for the length of the *49er* track, you assume he wants to record the entire song.

Once these tasks are completed the assignment is over. However, you could combine the two separate sound tracks into one track via the use of the Transport module as described in the Transport Reference Section. This would make re-creation of the sound easier and may be necessary if you have a slow hard disk.

Index