Chapter 4 ✍ StudioCard Operational Guidelines

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Introduction

The purpose of this chapter is to assist you in configuring and optimizing the *StudioCard*'s settings, including:

- ✍ Configuring Clock Source
- ✍ Setting Levels
- ✍ Effectively Using Digital Feedthrough

Levels and Headroom

The purpose of this section is to explain signal levels and headroom as they pertain to the *StudioCard* so that you can make recordings with optimum quality and signal-to-noise ratio. The information is this section is written assuming you are familiar with decibels: a unit of amplitude measurement commonly used when working with audio. Many sources exist describing decibels, including information on the Antex web page, www.antex.com.

The Numbers

The balanced input and output signal levels of the *StudioCard* are +4dBu nominal, +24dBu maximum. For unbalanced inputs, the signal levels are –10dBV nominal and 10dBV maximum. The difference between nominal and maximum level is referred to as headroom; the *StudioCard* is designed to have 20dB of headroom.

Headroom is required to handle peaks and transients in an analog signal to prevent the input signal from exceeding the range of the A/D converter. When this happens, sound quality degrades very rapidly, and is termed “digital clipping”.

These levels were selected for the *StudioCard* because SMPTE (Society of Motion Picture and Television Engineers) is working to standardize -20dB as the nominal recording level for all digital recording. A headroom figure of 20dB is more than many users are accustomed to, and has implications on volume and signal to noise performance as explained below.

Interpreting Input / Output Trim Levels

The *StudioCard* allows two input/output trim levels. They are set in the mixer and are marked +24 and +12.

- ✍ When +24dBu is selected, a standard nominal level of +4dBu (most balanced pro equipment) will produce a reading of -20dB on the level meter. This allows a very conservative 20dB before clipping.
- ✍ When +12dBu is selected, a standard nominal level of -10dBV (most home audio equipment) will produce a reading of -20dB on the level meter. Again this allows a very conservative 20dB before clipping. Playback levels will follow recording levels.

These levels require all level control in the mixer to be set at their full on positions (default). Levels can be lowered but not increased above these values.

A Little History

In analog tape recording the level meters are usually scaled in dBVu or Volume Units calibrated in decibels. At the 0Vu point the scale usually changes to red and is labeled in +dB units. We are warned that going into the red will cause distortion in the recording. This is true, but in analog tape recording this distortion can be tolerable well into the red region.

In digital audio recording there is an absolute maximum level that can be recorded and any signal above this level will be clipped (digital clipping). Clipping will produce intolerable amounts of distortion. All metering for digital audio places 0Vu at the clipping point. This means that you must keep your average and peak levels below the 0Vu point.

Maximizing Signal to Noise Ratio / Setting Levels

To maximize the Signal to Noise ratio of a recording, it is important to input the signal into the *StudioCard* at the proper level. The trim level should be set to match the levels of the equipment the *StudioCard* is connected to. A signal optimized such that its peaks are near 0dB on the meters in the Antex Demo will be recorded with peak signal to noise ratio.

It is suggested you use the amplitude control of your external equipment to input signals of the correct amplitude and headroom setting into the *StudioCard*. If you do not have adequate external adjustment and your input signal is of limited dynamic range, setting the trim level in the mixer to +12 will increase the amplitude of the recorded signal and therefore increase the Signal to Noise ratio. The amount of headroom will be reduced accordingly, and caution must be taken to verify digital clipping is avoided for large peaks.

Most commercial POP music CDs are recorded within 3 to 6dB of the 0Vu clipping point, while Classical CDs may average 20dB below 0Vu with large peak levels that come very close to 0Vu. The recordist must decide how much “headroom” to provide for the dynamic range of the music or sound being recorded.

Using the StudioCard with Video Capture & Display Boards

Audio sampling on most audio cards is generated by a crystal oscillator. Since this oscillator has no reference to the oscillator on the video board used to generate frames of video, the audio and video can drift out of synchronization. This is the equivalent of setting two watches to exactly the same time and watching them slowly drift apart. The *StudioCard* has the unique ability to lock its internal oscillator to timing signals provided by most video capture and display boards. When the *StudioCard* is referenced to this video timing signal it is impossible for audio to drift out of synchronization with video. This is extremely important when long video clips are used.

It is important to note that one of the locking methods listed below must be in place for **both** capture and playback of the source audio to insure synchronization. Audio digitally input to the *StudioCard* (S/PDIF or AES/EBU) will **not** be locked to the video as the sample clock used to digitize the audio was not locked to the video sample clock.

Synchronizing the StudioCard

Described in this section are two methods of synchronizing the *StudioCard* to the PVR, Truevision, Intergraph, Matrox, and other video capture boards:

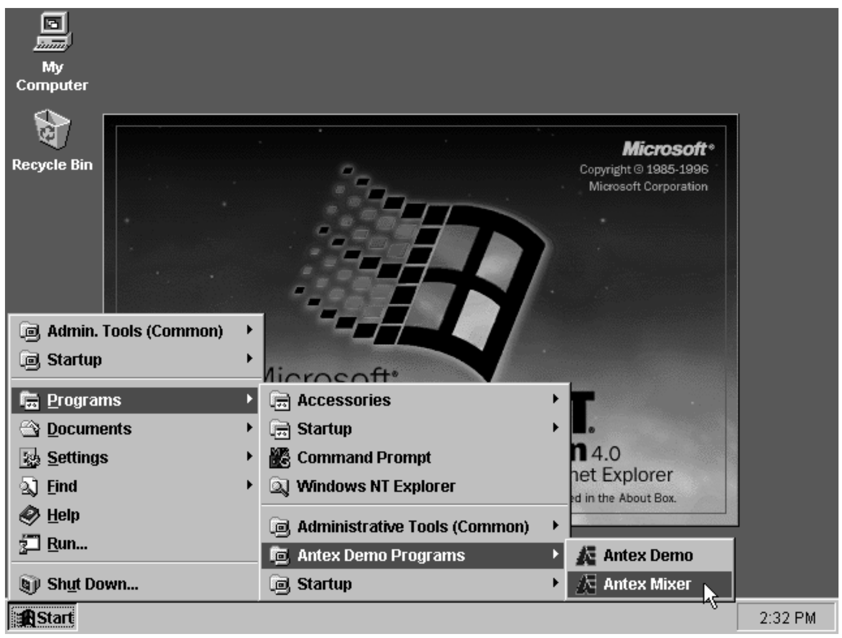
- ✍ Master 27 MHz clock lock (PVR only)
- ✍ Horizontal sync lock (all video boards)

Master 27 MHz clock lock (PVR only)

This method can only be used with the Perception Video Recorder from DPS. It works only when the Perception Effects (PFx) board is **not installed**.

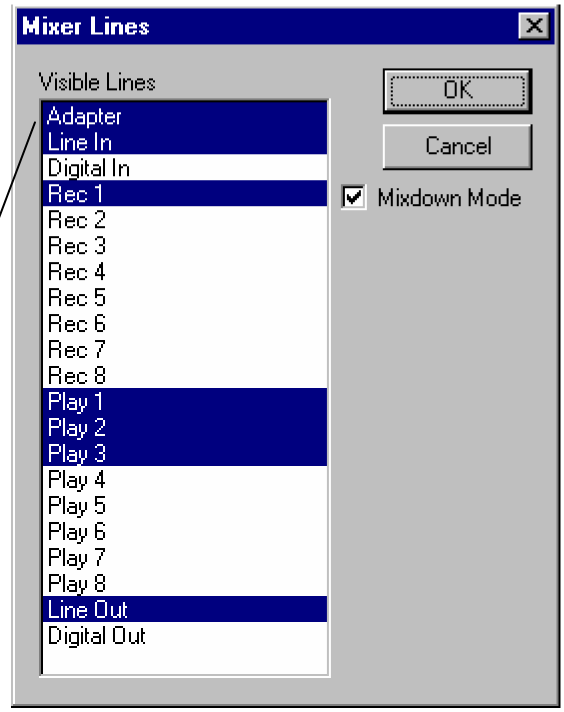
After installing the PVR sync cable (as described in Chapter 1), follow the instructions below.

Step 1 Start or reboot the computer, then start Antex Mixer.

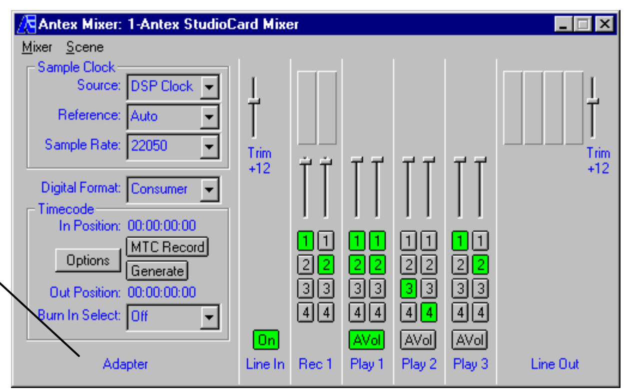


Step 2

If the Adapter line is displayed, continue to the next step. Otherwise, select **Lines…** from the **Mixer** menu. In the Mixer Lines window highlight **Adapter** and click OK.

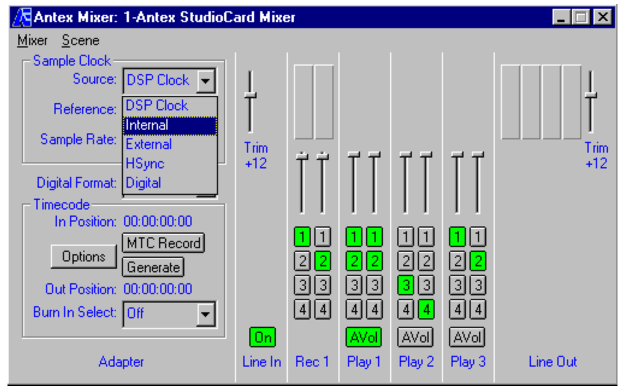


The Mixer main screen will display the Adapter line shown at right.



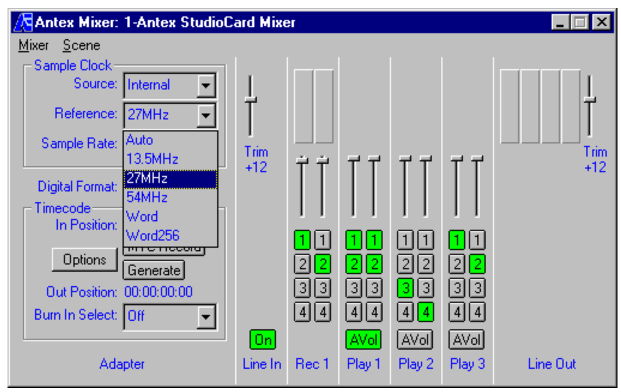
Step 3

Select **Internal** as the Sample Clock **Source**.



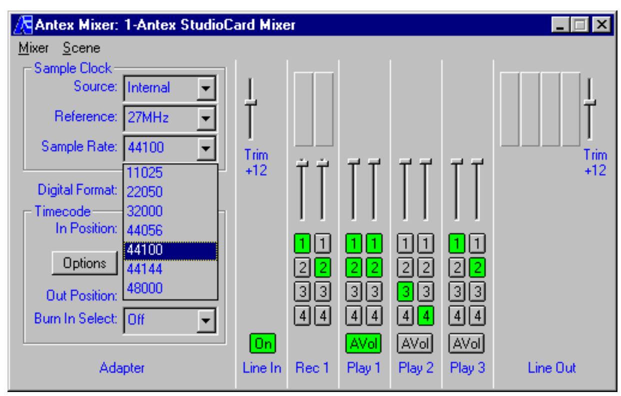
Step 4

Select **27MHz** as the Sample Clock **Reference**.



Step 5

If you are using SpeedRazor Mach 3.x, select **44100** as the Clock **Sample Rate**. Other sample rates can be selected as needed for other applications.



Note *This method will not work if you are using the Perception Effects board (Pfx). Use the following method if you have the Pfx board installed.*

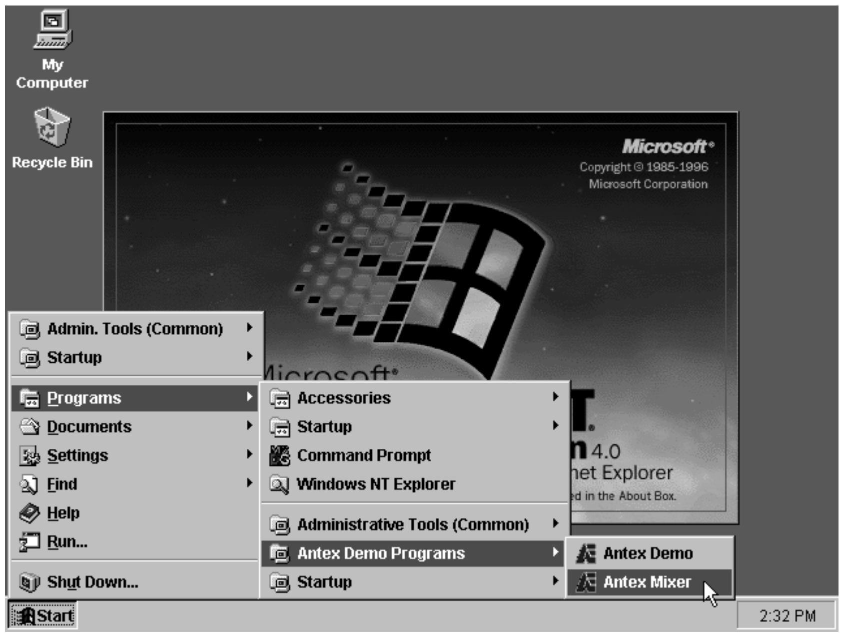
Horizontal sync lock (all video boards)

This type of lock if often referred to as gen-lock in video circles. With this method the *StudioCard* is locked to the horizontal synchronizing pulses that are part of the video signal. The *StudioCard* will auto-detect either NTSC or PAL video inputs.

To use this type of lock you must connect the composite video output signal from your video board to the video input of the *StudioCard*, as described in Chapter 1. The composite video signal is looped through the *StudioCard* and appears on the video output connector. This can be used to feed another video device such as a VTR.

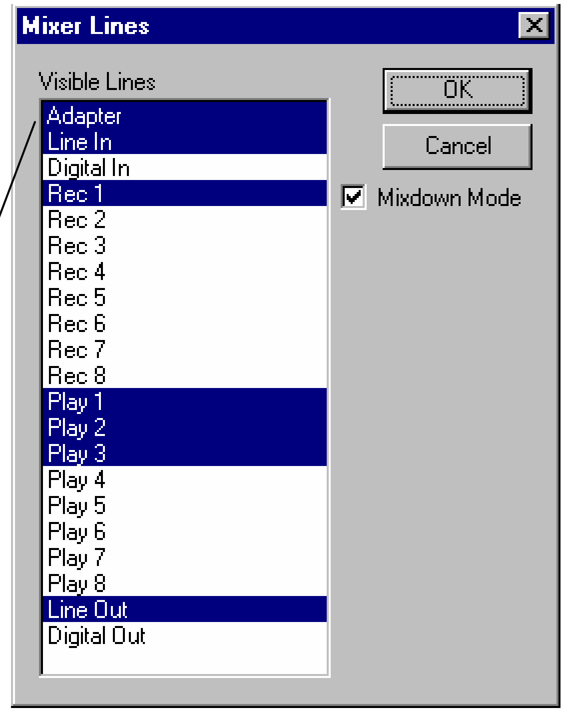
Follow the instructions below.

Step 1 Start Antex Mixer.



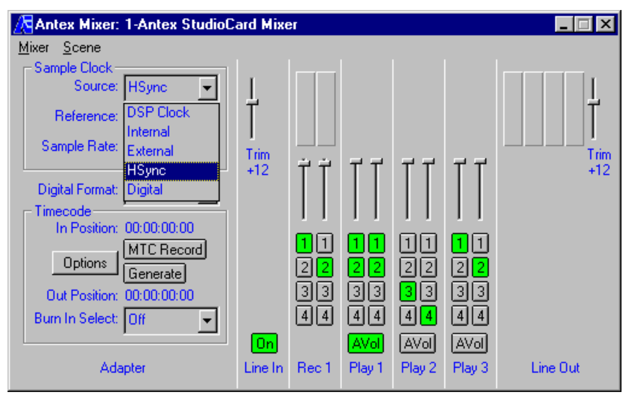
Step 6

If the Adapter line is displayed, continue to the next step. Otherwise, select **Lines…** from the **Mixer** menu. In the Mixer Lines window highlight **Adapter** and click OK.



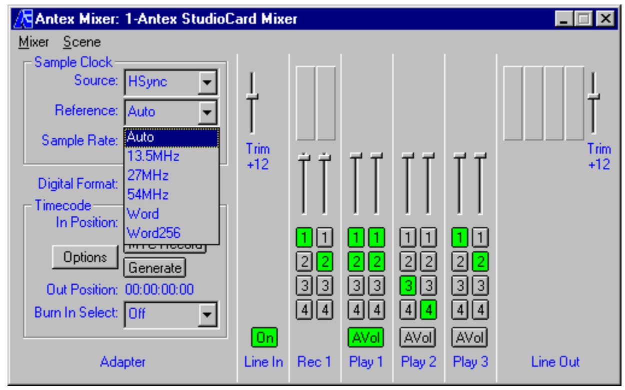
Step 7

Select **Hsync** as the Sample Clock **Source**.



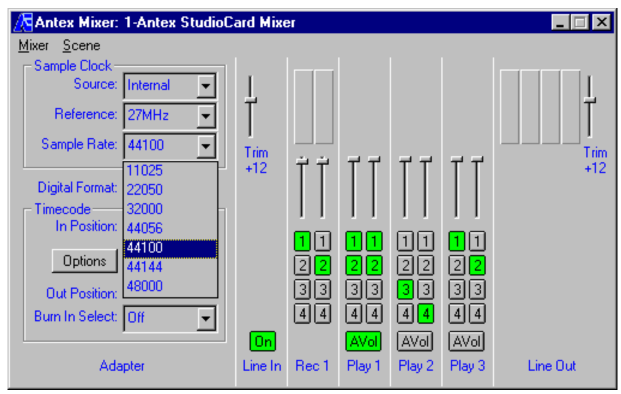
Step 8

Select **Auto** as the Sample Clock **Reference**.



Step 9

If you are using SpeedRazor Mach 3.x, select **44100** as the Clock **Sample Rate**. Other sample rates can be selected as needed for other applications.



Monitoring / Feedthrough

"Monitoring" is a term used to describe listening to the signal that is being recorded. It is also referred to as "Feedthrough" audio, as the input signal is fed to an output line. Feedthrough audio is often used to set the levels on input signals to optimize their amplitude and signal to noise ratio. Two types of feedthrough are possible, analog and digital. This section will explain the differences, implementation on the *StudioCard*, and how to enable and use it effectively.

- ✍ Analog Feedthrough routes the analog signal at the input channel directly to the output channel. In this manner, the signal bypasses the Analog to Digital (A/D) and Digital to Analog (D/A) converters and their associated filter and gain stages. This is very convenient at times, but does not monitor the actual signal being recorded. For example, if the signal amplitude is brought in too large (hot), the A/D may be in digital clipping (and sound awful) while the analog Feedthrough sounds fine.

- ✍ The second method is digital Feedthrough, the type supported by the Antex *StudioCard*. This method routes the digitized analog input signal at the output of the A/D directly to the D/A. The output of the D/A is filtered, amplified, and routed to one of the analog outputs. This approach is exactly analogous to a two head audio tape machine that allows the play head to play data immediately after the record head places the data on the tape.

The advantage of digital feedthrough is that the monitored signal is based on the digital data being recorded to the computer's hard disk. What you hear is truly what you get. By default, digital feedthrough is enabled on the *StudioCard*, enabling you to hear audio during recording. Record device one is mapped to playback device one and so on. Enabling and disabling digital feedthrough is accomplished by following the *Simultaneous Record-Play (SRP) Mode* instructions in Chapter 3, *Using the Antex Mixer*.

Digital feedthrough is a very convenient feature, but does require the use of an application program to place the *StudioCard* in record (or pause) mode. The Antex Demo application is one simple and easy to use tool to accomplish this. The Demo program also allows you to place the input in record/pause mode (again, the same as an analog tape deck). The input signal may be monitored and its levels adjusted without writing any data to the disk. Using the Antex Demo in this manner is described in the *Digital Feedthrough Mode* section of Chapter 2, *Using the Antex Demo*.

