Nuendo 96/52 DSP

**PCI Busmaster Digital I/O Card
2 + 24 Channels Stereo / ADAT Interface
24 Bit / 96 kHz Digital Audio
ADAT Sync In
MIDI I/O**

User Manual

Contents

Introduction

Thank you for choosing the Nuendo 96/52 DSP. This unique audio system is capable of transferring digital audio data directly to a computer from practically any device equipped with a digital audio interface, be it S/P-DIF, AES/EBU or ADAT optical. The numerous unique features and well thought-out Settings dialog positions the Nuendo 96/52 DSP at the very top of the range of digital audio interface cards. Installation is simple, even for the Windows novice, thanks to the latest Plug and Play technology and full interrupt-sharing.

The package includes drivers for Windows 98/2000/XP and Mac OS.

Our high-performance philosophy guarantees maximum system performance: All functions are directly executed by the hardware and not by the driver (i.e. the CPU).

Package Contents

Please check that your Nuendo 96/52 DSP System's package contains each of the following:

- Nuendo 96/52 DSP PCI card
- Nuendo 96/52 DSP expansion board
- User Manual
- Driver CD
- Adapter cable (D-type to D-type/phono)
- MIDI breakout cable
- Internal cable (2-core)
- Flat ribbon cable (14-core)
- 2 optical cable (TOSLINK), 2 m (6.6 ft)

System Requirements

Windows PC

- Windows 98/SE/ME, Windows 2000/XP
- PCI Interface: a free PCI rev. 2.1 Busmaster slot
- CardBus Interface: a free PCMCIA Slot type II, CardBus-compatible

Macintosh-Computer

- Mac OS X 10.2, running on a compatible computer.
- PCI Interface: a free PCI rev. 2.1 Busmaster slot
- CardBus Interface: a free PCMCIA Slot type II, CardBus-compatible

Brief Description and Characteristics

- 0 percent (zero!) CPU load, even when all 52 ASIO channels are used
- All settings can be changed in real-time
- Enhanced mixed mode: ADAT In, S/PDIF In, and all outputs can be used simultaneously
- 8 available buffer sizes/latencies: 1.5 / 3 / 6 / 12 / 23 / 46 / 93 / 186 ms
- Sample Split technology for 12 channel, 96 kHz/24-bit record/playback via ADAT optical
- Slave and Master clock modes
- Automatic and intelligent Master/Slave clock control
- Unsurpassed Bitclock PLL (audio synchronization) in ADAT mode
- Word clock input and output
- ADAT Sync in (9-pin D-type) for sample-accurate transfer
- Zero Latency Monitoring: Hardware bypass per track, controlled by Punch in/out
- Latency-free submixes and perfect ASIO Direct Monitoring
- Sample aligned and never swapping channels
- SyncCheck tests and reports the synchronization status of input signals
- 2 x MIDI I/O, 32 channels high-speed MIDI
- DSP Mixer: 1352 channel mixer with 40 bit internal resolution

Technical Specifications

Digital

- Low jitter S/P-DIF: < 3 ns in PLL mode (44.1 kHz, optical in)
- Low jitter ADAT: < 2 ns in PLL mode (44.1 kHz, optical in)
- Input PLL ensures zero dropout, even at more than 40 ns jitter
- Bitclock PLL for trouble-free varispeed operation in ADAT mode
- High-sensitivity input stage (< 0.2 Vss input level)
- Output voltage 0.8V (consumer mode, phono) or 2.3V (professional mode)
- Sample frequencies: 32 / 44.1 / 48 / 88.2 / 96 kHz and variable

Digital Interface

- Phono input and output ground-free transformer coupled
- Connectors: optical (TOSLINK), phono, BNC
- Clocks: ADAT Sync In, Word clock I/O
- Formats: S/P-DIF (Consumer and Professional), ADAT optical

MIDI

- 2 x MIDI I/O via breakout cable
- PCI bus based hi-speed operation
- Separate 128 byte FIFO for input and output
- MIDI state machine in hardware for reduced interrupt request load

Transfer Modes: Resolution / Bits per Sample

Windows PC

ASIO:

- 24 or 32 bit, 4 byte (stereo 8 byte)

  This format is compatible with 16-bit and 20-bit. Resolutions below 24-bit are handled by the audio application.

MME:

- 16 bit, 2 byte (stereo 4 byte)
- 20 bit, 3 byte MSB (stereo 6 byte)
- 20 bit, 4 byte MSB (stereo 8 byte)
- 24 bit, 3 byte (stereo 6 byte)
- 24 bit, 4 byte MSB (stereo 8 byte)
- 32 bit, 4 byte (stereo 8 byte)

Macintosh Computer

- 32 bit, 4 byte (stereo 8 byte)

  This format is compatible with 16-bit and 20-bit. Resolutions below 24-bit are handled by the audio application. Internally, the card uses 32-bit format, but audio data transfer is limited to 24-bits.

Hardware Installation

❐ **Before installing the PCI card, please make sure the computer is switched off and the power cable is disconnected from the mains supply. Inserting or removing a PCI card while the computer is in operation can cause irreparable damage to both motherboard and card!**

1. Disconnect the power cord and all other cables from the computer.

2. Open the computer's housing. Further information on how to do this can be obtained from your computer´s instruction manual.

3. Important: Before removing the Nuendo 96/52 DSP from its protective bag, discharge any static in your body by touching the metal chassis of the PC.

4. Connect the Nuendo 96/52 DSP card with the Expansion Board using the supplied flat ribbon cable. Note: The connectors on the cable cannot be plugged in the wrong way round.

5. Insert the Nuendo 96/52 DSP firmly into a free PCI slot, press and fasten the screw.

6. Insert the Expansion Board and fasten the screw.

7. Close the computer's housing.

8. Reconnect all cables including the power cord.

• **Note: If neither Word clock I/O, MIDI I/O nor a third ADAT I/O is required, it is not necessary to install the Expansion Board at all (i.e. leave out steps 4 and 6).**

Driver Installation

Windows 98/SE/ME

If the PCI card has been installed correctly (see "Hardware Installation" on page 9), and the computer has been switched on, Windows will recognize the new hardware component and start its "Add New Hardware Wizard". Insert the Driver CD into your CD-ROM drive and follow the instructions displayed on your computer screen. During installation, locate the driver files in the respective directory on the Driver CD.

Windows will install the Nuendo 96/52 DSP System driver and register the card in the system as a new audio device. The computer should now be re-booted.

❒ **Sometimes the CD-ROM path (i.e. its drive-letter) must be typed in again manually during the copy process.**

Using the Nuendo 96/52 DSP's Settings dialog (see page 24), you can easily configure the Nuendo 96/52 DSP system.

Windows 2000/XP

If the PCI card has been installed correctly (see "Hardware Installation" on page 9), and the computer has been switched on, Windows will recognize the new hardware component and start its "Hardware Wizard". Insert the Driver CD into your CD-ROM drive and follow the instructions that appear on your computer screen. During installation, locate the driver files in the respective directory on the Driver CD.

Windows will install the Nuendo 96/52 DSP System driver and register the card in the system as a new audio device. After a reboot the Nuendo 96/52 DSP is ready for use.

Using the Nuendo 96/52 DSP's Settings dialog (see page 24), you can easily configure the Nuendo 96/52 DSP system.

❒ **You can safely ignore alert messages like "Digital signature not found", "Do not install driver", "Not certified driver". Simply continue with the installation.**

Macintosh Computer

❒ **If an older version of the driver is already installed, first make sure to remove all old files. To do so open the Extensions folder that resides in your System folder. Drag the old driver onto the Trash to delete it. Also delete the Nuendo 96/52 DSP Settings file.**

After you have installed the hardware (see "Hardware Installation" on page 9) and restarted your computer, you can install the drivers from the Driver CD.

If you have downloaded a newer version from the Steinberg website, double click on the archive file to unpack it.

After unpacking the driver files reside in a virtual drive on your desktop. The Installation process is identical, no matter if you install from here or from CD.
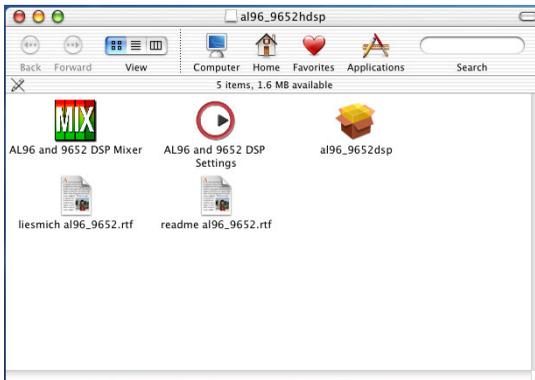
Open the CD or virtual drive window.

You can copy the files "AL96 and 9652 DSP Mixer" and "AL96 and 9652 DSP Settings" to any desired location on your hard drive. You can e.g. create a special folder for them.

Then drag these two files from the CD or virtual drive window onto the desired location, e.g. into the newly created folder.

After the installation of the driver files and a computer reboot, as described hereafter, you can double click on the files that you just dragged, to either bring up the card's DSP Mixer or its Settings dialog.

Start driver file installation by double clicking on the file "al96_9652dsp"

The following Introduction dialog is displayed.

Click on the "Continue"button.
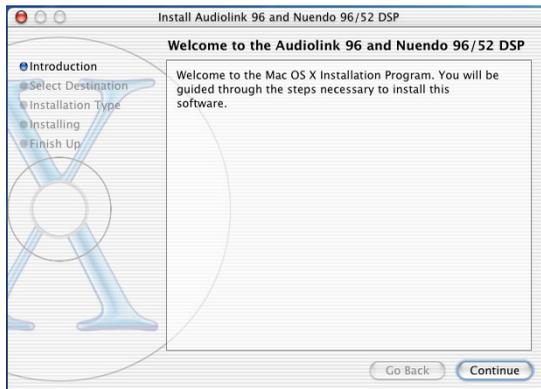
This dialog is used to select the drive that holds the System that you intend to use. Click on the desired drive icon. Then confirm by clicking "Continue".
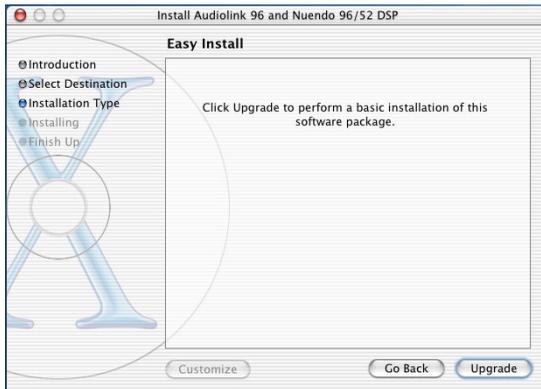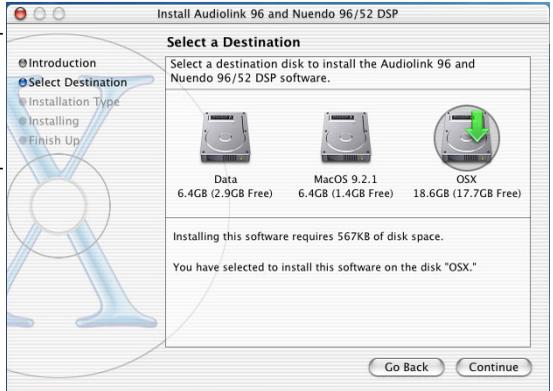
When you have completed these preparatory steps, you can now start the actual installation process from this dialog.

Do so.

When the installation process has finished, this dialog is shown.

You must now restart your computer to enable the computer to load the newly installed drivers and make them available to the System.

Setting the Nuendo 96/52 DSP is done in its Settings dialog. This can e.g. be started from within any ASIO compatible software (in Nuendo you would e.g. open the Device Setup dialog from the Devices menu, select VST Multitrack and click onto the Control Panel button).

Operation and Usage

External Connections

Nuendo 96/52 DSP consists of the main PCI board and an Expansion Board. All the essential electronics are located on the PCI card, so it will also work without the Expansion Board.
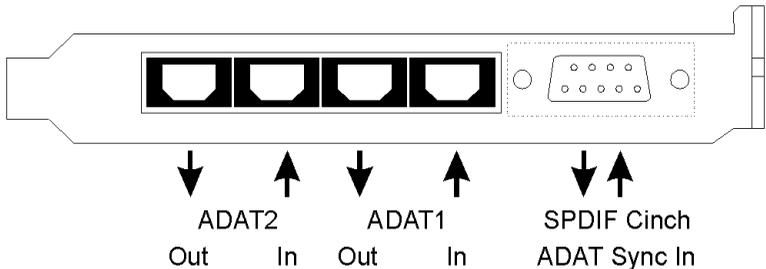
The main board's bracket has two ADAT optical inputs and two ADAT optical outputs, as well as a 9-pin D-type socket. Coaxial S/PDIF input and output requires plugging in the adapter cable, whereby the red phono socket is the output. The ADAT1 I/O next to the D-type socket can also be used for optical S/P-DIF, if this mode is selected in the Settings dialog.

S/P-DIF configuration is done in the Settings dialog, which may e.g. be opened from within the respective ASIO host application (in Nuendo and Cubase SX you will find it under the path Devices menu/Device Setup/Control Panel button. The Nuendo 96/52 DSP card accepts all common digital audio sources; S/P-DIF as well as AES/EBU. Channel status and copy protection are ignored.

In S/P-DIF mode, identical signals are available at both the optical and the coaxial outputs. An obvious use for this would be to connect two devices, thus using the Nuendo 96/52 DSP as a splitter.

To receive signals in AES/ EBU format, an adapter cable is required. Here Pins 2 and 3 of a female XLR plug must separately be connected to the two pins of a phono plug. The cable shielding may only be connected to pin 1 of the XLR - *not* to the phono plug.
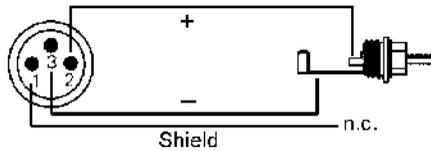
The ground-free design using transformers for digital inputs and outputs enables trouble-free connection to all devices, and perfect hum rejection.
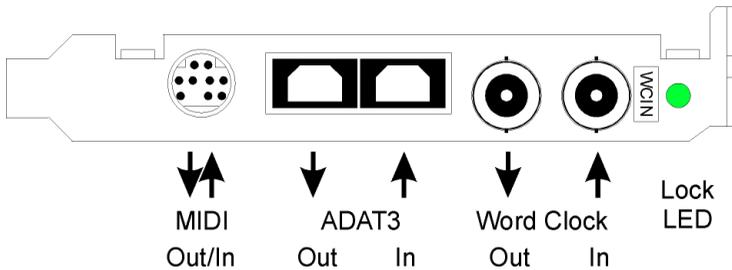
The Expansion Board's bracket provides a third ADAT optical input and output as well as Word clock I/O. Next to the two BNC sockets is an LED, that displays the Word clock input lock status. When you connect tje included breakout cable to the 9-pin Mini-DIN connector, it provides two MIDI inputs and outputs.

Internal Connections

The internal digital input (connector ST3 on the circuit board, CD IN) can be connected to the digital output of an internal CD-ROM drive. This allows for direct transfer of digital audio data within the computer.

The connectors SYNC1, SYNC2 and ST9 have no function at this time. They are reserved for future enhancements.

X4 is a connector used to program the card in the factory.

Windows MME Playback

The Nuendo 96/52 DSP system can only play back audio data in the formats (sample rate, bit resolution) that it supports. If an unsupported format is used, an error message will appear (for example at 22 kHz and 8 bit).

First, you must select the Nuendo 96/52 DSP as output device in your audio application. This can often be done in the Options, Preferences or Settings menus using functions that may be named Playback Device, Audio Devices, Audio etc. To make full use of the Nuendo 96/52 DSP's potential, we recommend that you use 24-bit resolution for playback.

We strongly recommend that you switch all system sounds off:

• Under Windows 98/SE/ME you can reach the appropriate dialog from the Start menu (via Settings/Control Panel/Sounds).
• Under Windows XP these settings can be accessed from the Start menu via Settings/Sounds, selecting the Audio Devices Sounds tab and opening the respective pop-up menu.

Nuendo 96/52 DSP should not be selected as the "Preferred or Standard Device" for system sound playback, as this could cause loss of synchronization and unwanted noise.

• Under Windows 98/SE/ME you will find the respective parameter under Settings/Control Panel/Multimedia/Audio tab, Preferred Device pop-up.
• Under Windows XP you must open the Start menu and use the path Settings/Sounds and Audio Devices/Audio tab, Standard Device pop-up to make the corresponding setting.

If you feel that you can not do without system sounds, you should consider buying a cheap Blaster clone and then select this as Preferred/Standard Device.

The illustration below shows a typical configuration dialog as used by a standard (stereo) wave editor. After selecting a device, audio playback is sent either to S/P-DIF or to the ADAT ports, depending on which has been selected as playback device.
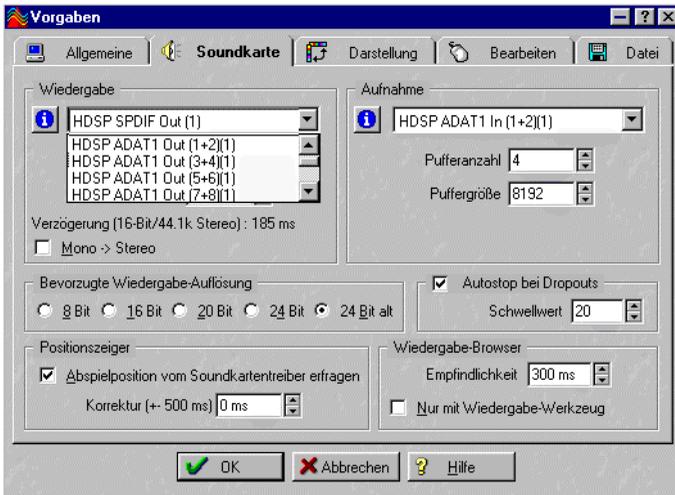
Increasing the number and/or size of audio buffers may prevent the audio signal from breaking up, but also increases latency i.e. output is delayed. For synchronized playback of audio and MIDI (or similar), be sure to activate the checkbox "Get position from audio driver". Even at higher buffer settings, in a mixed Audio/MIDI environment, sync problems should not occur as the Nuendo 96/52 DSP always reports the correct current play position (even while recording – essential for chase lock synchronization).

The Nuendo 96/52 DSP system's ADAT optical interface allows sample rates of up to 96 kHz using a standard ADAT recorder. At this frequency a "Sample Split" method is used: single-channel data is distributed onto two ADAT channels. This reduces the number of available ADAT channels. Instead of previously 24 only 12 channels are available.

Under Windows MME and in double-speed mode, channels are routed to ADAT devices as follows:

• Only the stereo pairs (1+2) and (3+4) of each ADAT port are available

- Channel 1 is routed to channels 1 and 2, channel 2 is routed to 3 and 4 etc.

Please refer to the diagram "ADAT Track Routing, MME 96 kHz" in the "Diagrams" section that starts on . Record and playback routing is identical.

Windows MME Recording

Unlike analog soundcards, which produce empty wave files (or noise) when no input signal is present, digital I/O cards always need a valid input signal to start recording.

To take this into account, the Nuendo 96/52 DSP system offers two unique features: a comprehensive I/O signal status display (showing sample frequency, lock and sync status) in the Settings dialog and the protective Check Input function.

If a 48 kHz signal is fed into the input and the application is set to 44.1 kHz, Check Input will prohibit recording. This is designed to prevent faulty takes, which may not sound bad, but use the wrong playback sample rate. This error would otherwise often go unnoticed until a later production stage.

The sample frequency shown in the Settings dialog (see the illustration on) offers a quick overview of the current configuration (of the card itself and all connected external equipment). If no sample frequency is recognized, a "No Lock" message is displayed in the dialog.

This makes it simple to configure any suitable audio application for digital recording. When you select the required input, Nuendo 96/52 DSP displays the current sample frequency. This parameter can then be changed in the audio application's Audio Attributes (WaveLab) or a similar dialog.

The screenshot to the right shows a typical dialog (taken from WaveLab) used for changing basic parameters such as sample frequency and resolution in an audio application.
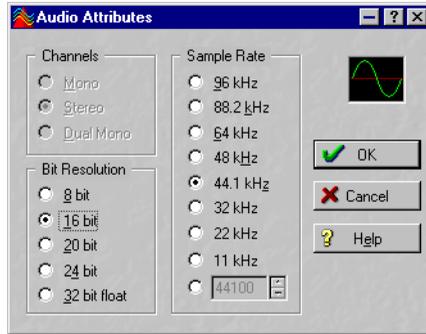
Any bit resolution can be selected, providing it is supported by both the audio hardware and the software.

Even if the input signal is 24 bit, the application can still be set to record at 16-bit resolution. The lower 8 bits (and therefore any signals circa 96dB below maximum level) are lost entirely. On the other hand, there is nothing to gain from recording a 16-bit signal at 24-bit resolution - this would only waste precious hard disk space.

It often makes sense to monitor the input signal or send it directly to the output. This can be done at zero latency using DSP Mixer (see "DSP Mixer: Routing and Monitoring" on page 50).

Nuendo 96/52 DSP also includes a useful automatic real-time monitor function, see "Monitor – A Function of the Windows MME Driver" on page 60 for details. Activating record or record-ready status in the application causes the input signal to be routed according to the current mixer settings.

Two solutions are currently available to enable an automated control of real-time monitoring.

• Steinberg's ASIO protocol with the Nuendo 96/52 DSP ASIO 2.0 drivers and any ASIO 2.0 compatible program. When "ASIO Direct Monitoring" is active and Recording starts, the input signal will be routed to the output in real-time.
• ZLM (Zero Latency Monitoring) allows monitoring in Punch I/O mode. In this mode the system behaves like a tape machine. This method has been implemented in all versions of Samplitude (by SEK'D) and can be activated using the global track option 'Hardware monitoring during Punch'.

DVD-Playback (AC-3/DTS) under MME

Via the Nuendo 96/52 DSP's S/P-DIF output, popular DVD software players like WinDVD and PowerDVD can send their audio data stream to any receiver capable to digest AC-3/DTS. For this to work, the S/P-DIF output wave device of the Nuendo 96/52 DSP must be selected in "Control Panel/Sounds and Multimedia/Audio" under Windows 98/SE/ME. Also activated "use preferred device only".

If you use Windows XP, follow the path Settings/Sounds and Audio Devices/Audio and select the S/P-DIF output wave device of the Nuendo 96/52 DSP there.

You will notice that "S/P-DIF Out" or "activate S/P-DIF output" have now become available in the DVD software's audio properties. When you select one of these, the software will play back the non-decoded digital multichannel data stream via the Nuendo 96/52 DSP card.

❐ **This "S/P-DIF" signal sounds like chopped noise at maximum level. It is therefore essential that you check "Non-audio" in the card's Settings dialog. Most S/P-DIF receivers will then not process the signal directly, which prevents damage to any attached equipment.**

Setting the card so that it's used as system playback device is against our previous recommendations as professional cards shouldn't be hampered by system events and are not specialized to play back system sounds. After DVD playback is over, you should therefore make sure to re-assign the previous settings. As an alternative, you could also disable any system sounds (Sounds tab, "No audio" setting).

• **Note: The DVD player will be synced from the Nuendo 96/52 DSP card. This means if AutoSync and/or Word clock are used, playback speed and pitch will change according to the incoming clock signal/sample frequency.**

Low Latency under MME (Buffer Size Adjustment)

Under Windows 95 or 98 the MME buffer size was nothing to worry about. Latencies below 46 ms were not possible. Meanwhile both computers as well as the operating system have become much more powerful, and since the advent of Windows ME/2000/XP far lower latency values can be used under MME. Among the applications that support such small latency values are WaveLab version 3.04 or newer, Sequoia version 5.91 or higher as well as SAWStudio and Sonar.

In the Nuendo 96/52 DSP's Settings dialog, the MME buffer size (more precise: the DMA buffer size) is set with the same buttons as the ASIO buffer size. Our test computers allow settings down to 64 samples without causing clicks. Please note that this setting only defines the latency of the hardware. The true and effective latency is configured within the MME application!

❑ **Attention: the DMA buffers must not be larger than the application's buffers. This can easily happen when you use ASIO and MME at the same time (multi-client operation) and ASIO is set to 186 ms while the buffers in the MME application are still set for a lower latency. Audio playback will stutter and be distorted.**

• **Example: when you set the Nuendo 96/52 DSP to 512 you can't use 128 in any program. But setting DMA to 128 allows you to use 128 and all higher values within the software.**

Please also note that this is a "you're welcome to try" feature. We can't guarantee that you will be able to use 3 or 6 ms with MME. Simply try out the lowest settings that your system and software allow. Some motherboards with insufficient PCI bandwidth (especially VIA based) suffer from crackling noise at settings below 512. In such a case, make sure to set the buffer size to 512 or higher.

Macintosh Computer – Digital Recording

Unlike analog soundcards which produce empty wave files (or noise) when no input signal is present, digital I/O cards always need a valid input signal to start recording.

To take this into account, two unique features have been included in the Nuendo 96/52 DSP system: a comprehensive I/O signal status display (showing sample frequency, lock and sync status) in the Settings dialog and status LEDs for each input.

The sample frequency displayed in the Settings dialog (see the Settings dialog illustration on page 32) offers a quick overview of the current configuration (of the card itself and all connected external equipment). If no sample frequency is recognized, a "No Lock" message is displayed in the dialog.

This makes it simple to configure any suitable audio application for digital recording. When you select the required input, Nuendo 96/52 DSP displays the current sample frequency. This parameter can then be changed in the audio application's Audio Attributes (WaveLab) or a similar dialog.

It often makes sense to monitor the input signal or send it directly to the output. This can be done at zero latency using DSP Mixer (see "DSP Mixer: Routing and Monitoring" on page 50).

For automated real-time monitoring function the Nuendo 96/52 DSP System supports ASIO Direct Monitoring (ADM) in ASIO 2.0. If "ASIO Direct Monitoring" has been switched on the input signal is routed to the output in real-time whenever recording starts.

Nuendo 96/52 DSP Configuration – Windows PC

General Information

Configuring the Nuendo 96/52 DSP system is done using its own Settings dialog. This can be opened using two different methods:

- By clicking on the Steinberg dial icon in the Taskbar's system tray
- By using the corresponding link on the Desktop

Dial symbol

DSP-Mixer symbol
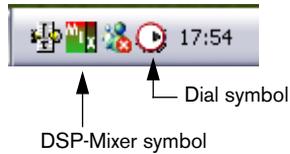
The DSP Mixer can also be opened in two different ways:

- By clicking on the MIX icon in the Taskbar's system tray
- By starting the corresponding link from the Desktop

The Nuendo 96/52 DSP's hardware offers a number of useful, well thought and practical functions and options which affect how the card operates. It can be configured to suit many different requirements. The following is available in the "Settings" dialog:

- Input selection
- Output mode
- Output channel status
- Synchronization behaviour
- Input and output status display
- Time code display

Settings-Dialog unter Windows XP.

Any changes made in the Settings dialog are applied immediately - confirmation (e.g. by clicking on OK or exiting the dialog) is not required. However, settings should not be changed during playback or record if it can be avoided, as this can cause unwanted noise.

Also, please note that even in "Stop" mode, several programs keep the recording and playback devices open, which means that any new settings might not be applied immediately.

The status displays at the bottom of the dialog box give the user precise information about the current status of the system and the status of all signals.

• SyncCheck indicates whether there is a valid signal for each input ("Lock" or "No Lock"), or if there is a valid and synchronous signal ("Sync").
• The "AutoSync Ref" display shows the input and frequency of the current sync source.
• "Time Code" displays time information received from the I/O-box ADAT Sync port. This is convenient for checking whether the system is running in time with the transmitting device (e.g. ADAT).

MME

Check Input verifies the current input signal against the settings in the record program. When de-activated a recording will always be allowed, even with non-valid input signals.

TMS activates the transmission of Channel Status data and Track Marker information of the S/P-DIF input. Both settings are valid for MME only, thus not applicable to ASIO.

AEB

This is used to activate/deactivate the Extension Board.

Buffer Size

The Buffer Size setting determines the latency between incoming and outgoing ASIO (under Windows 2000/XP also GSIF) data. It also affects system stability (see "Operation under ASIO 2.0" on page 43). Under Windows MME this setting determines the DMA buffer size.

S/P-DIF In

Defines the input for the S/P-DIF signal. "Coaxial" relates to the phono socket, "ADAT1" to the optical TOSLINK input ADAT1.

S/P-DIF Out

The S/P-DIF output signal is constantly available at the phono plug. If you select "ADAT1", it is also routed to the optical output ADAT1. For further details about the settings "Professional", "Emphasis" and "Non-Audio", please refer to "Special Characteristics of the S/P-DIF Output" on page 42.
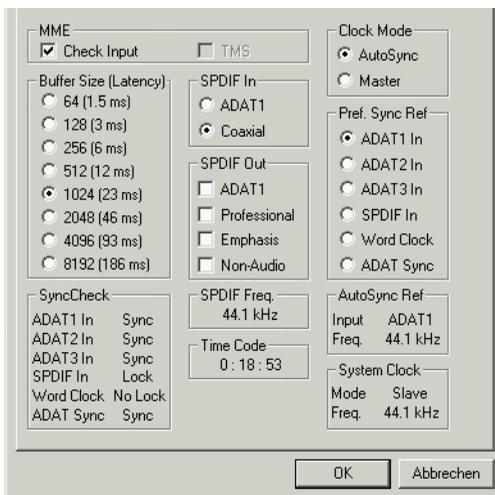
Settings dialog under Windows 98.

S/P-DIF Freq

In this section of the dialog, the S/P-DIF sample frequency is displayed.

Clock Mode

The card can be configured to use its internal clock (Master), or the clock source pre-defined via Pref. Sync Ref (AutoSync).

Pref. Sync Ref

This is used to pre-select the desired clock source. If the selected source isn't available, the card will change to the next available one. The currently used clock source and sample rate are displayed in the AutoSyncRef display.

The automatic clock selection checks and changes between the clock sources ADAT optical, S/P-DIF, Word clock and ADAT Sync. The latter is recommended especially for sample-accurate transfers under ASIO 2.0.

System Clock

This shows the current clock state of the Nuendo 96/52 DSP system. The system is either Master (using its own clock) or Slave (AutoSync Ref).

Clock Modes - Synchronization

In the digital world, all devices are either the "Master" (clock source) or a "Slave" that synchronizes to the Master. Whenever several devices are linked within a system, there must always be a single master clock. The Nuendo 96/52 DSP's intelligent clock control is very user-friendly, being able to switch between clock modes automatically. Selecting "AutoSync" will activate this mode.

In AutoSync mode, the system constantly scans all digital inputs for a valid signal. If this signal corresponds to the current playback sample rate, the card switches from the internal quartz (AutoSync Ref displays "Master") to a clock generated by the input signal (AutoSync Ref displays "Slave"). This allows on-the-fly recording, even during playback, without having to synchronize the card to the input signal first. It also allows immediate playback at any sample rate without having to reconfigure the card.

AutoSync guarantees that normal record and record-while-play will always work correctly. In certain cases however, e.g. when the inputs and outputs of a DAT machine are connected directly to the Nuendo 96/52 DSP, AutoSync may cause feedback in the digital carrier, so that synchronization breaks down. To prevent this, switch the Nuendo 96/52 DSP's clock mode over to "Master".

❐ **Remember that a digital system can only have one Master!**
If the Nuendo 96/52 DSP's clock mode is set to "Master", all other devices must be set to "Slave".

All the ADAT optical inputs in the Nuendo 96/52 DSP as well as the S/P-DIF input work simultaneously. However, as there is no input selector, the Nuendo 96/52 DSP must be informed which of the signals it should recognize as the sync reference (a digital device can only be clocked from a single source). This is why the system has been equipped with an automatic clock source selection, that adopts the first available input with a valid digital signal as the

Settings-Dialog unter Windows XP.

clock reference input. The input currently used as sync reference and its sample frequency are shown in the AutoSync Ref status field.

A preferred input can be defined via Pref. Sync Ref (Preferred Synchronization Reference). As long as the card recognizes a valid signal there, this input will be designated as the sync source. Otherwise the other inputs will be scanned in turn. If none of the inputs are receiving a valid signal, the card automatically switches clock mode to "Master".

Setting a "Pref Sync Ref" is essential. to cope with some situations which may arise in studio practice. One example: An ADAT recorder is connected to the ADAT1 input (ADAT1 immediately becomes the AutoSync source) and a CD player is connected to the S/P-DIF input. Try recording a few samples from the CD and you will be disappointed. Few CD players can be synchronized. The samples will inevitably be corrupted, because the signal from the CD player is read with the (wrong) ADAT clock i.e. it is out of sync. In this case, "Pref Sync Ref" should therefore be temporarily set to S/P-DIF.

If several digital devices are to be used simultaneously in a system, they not only have to operate with the same sample frequency but also be synchronous with each other. This is why digital systems always need a single device defined as "Master", which sends the same clock signal to all the other ("Slave") devices. The SyncCheck technology lets you eassly test and display the current clock status. The "SyncCheck" field indicates whether no signal ("No Lock"), a valid signal ("Lock") or a valid and synchronous signal ("Sync") is present at each of the digital clock source inputs. The "AutoSync Ref" display shows the current sync source and the measured frequency.

In practice, SyncCheck gives you a very quick overview whether all digital devices connected to the system are configured properly. With SyncCheck, finally everyone can master this common source of error, previously one of the most complex issues in the digital studio world.

An example to illustrate this: The ADAT1 and ADAT2 inputs receive signals from a digital mixing desk that has been set to clock mode "Internal" or "Master". An ADAT recorder is connected to the ADAT3 input. The Nuendo 96/52 DSP is set to AutoSync mode. As expected, SyncCheck shows that the ADAT1 and ADAT2 inputs are in sync (as they are driven by the same clock from the mixing desk), but shows "Lock" instead of "Sync" for the ADAT3 input. As the ADAT recorder does not receive any signals from the Nuendo 96/52 DSP or from the mixer, it generates its own clock at a rate that is (almost) the same as the sample frequency of the mixing desk – but not identical.

Solution: To drive the ADAT recorder from its digital input, set it to Slave mode (DIG) and connect the input to the Nuendo 96/52 DSP's ADAT3 output. The Nuendo 96/52 DSP is already in sync with the mixing desk, so it will send an identical (synchronous) signal to ADAT3 out. The ADAT recorder will lock onto this, its output will also be in sync. The signal from the ADAT recorder is now fully in sync with the signals from the mixing desk.

Thanks to its AutoSync technology and lightning fast PLLs, the Nuendo 96/52 DSP is not only capable of handling standard frequencies. It can also handle any sample rate between 25 and 105 kHz. Even the Word clock input, which most users will use in varispeed operation, allows any frequency between 25 kHz and 105 kHz.

At 88.2 or 96 kHz the sample frequency shown in the "S/P-DIF In" field differs from the one shown in "AutoSync Ref" if one of the ADAT inputs has been selected in "Pref Sync Ref". The card automatically switches to its Sample Split mode here, as ADAT optical inputs and outputs are only specified up to 48 kHz. Data from/to a single input/output is distributed onto two channels, the internal frequency remains at 44.1 or 48 kHz. In this case, the ADAT sample frequency is only half the S/P-DIF frequency.

Nuendo 96/52 DSP Configuration – Macintosh

General Information

Configuring the Nuendo 96/52 DSP system is done using its own Settings dialog, the program Nuendo 96/52 DSP-Settings.

The Nuendo 96/52 DSP hardware offers a number of helpful, well thought-out, practical functions and options which affect how the card operates. It can be configured to suit many different requirements. The following is available in the 'Settings' dialog:

- Input selection
- Output mode
- Output channel status
- Synchronization behaviour
- Input and output status display
- Time code display
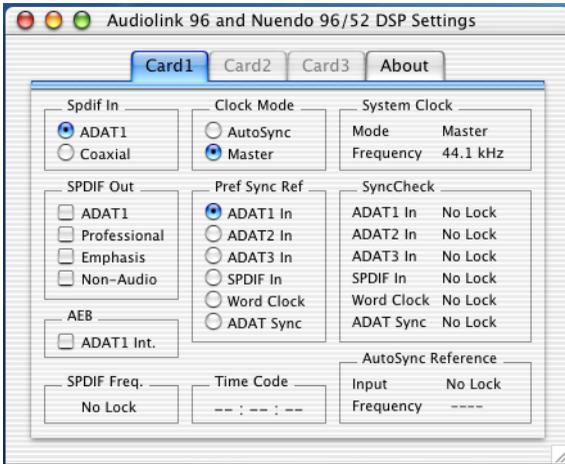
Any changes that you make in the Settings dialog are applied immediately – a confirmation (e.g. by clicking on OK or exiting the dialog) is not required. However, settings should not be changed during playback or record if it can be avoided, as this can cause unwanted noise. Also, please note that even in 'Stop' mode, several programs keep the recording and playback devices open, which means that any new settings might not be applied immediately.

The status displays at the bottom of the dialog box give the user precise information about the current status of the system and the status of all signals.

- SyncCheck indicates whether there is a valid signal for each input ("Lock" or "No Lock"), or if there is a valid and synchronous signal ("Sync").
- The "AutoSync Ref" display shows the input and frequency of the current sync source.
- "Time Code" displays time information received from the I/O-box ADAT Sync port. This is convenient for checking whether the system is running in time with the transmitting device (e.g. ADAT).

S/P-DIF In

Defines the input for the S/P-DIF signal. 'Coaxial' relates to the phono socket, 'ADAT1' to the optical input ADAT1.

S/P-DIF Out

The S/P-DIF output signal is constantly available at the phono plug. After selecting 'ADAT1' it is also routed to the optical output ADAT1. For further details about the settings 'Professional', 'Emphasis' and 'Non-Audio', please refer to .

AEB

This is used to activate/deactivate the Extension Board.

S/P-DIF Freq

In this section of the dialog, the S/P-DIF sample frequency is displayed.

Clock Mode

The card can be configured to use its internal clock (Master), or the clock source pre-defined via Pref. Sync Ref (AutoSync).

Pref. Sync Ref

This is used to pre-select the desired clock source. If the selected source isn't available, the card will change to the next available one. The currently used clock source and sample rate are displayed in the AutoSyncRef display.

The automatic clock selection checks and changes between the clock sources ADAT optical, S/P-DIF, Word clock and ADAT Sync. The latter is recommended especially for sample-accurate transfers under ASIO 2.0.

Buffer Size

The setting 'Buffer Size' determines the latency between incoming and outgoing data, as well as affecting system stability. We recommend selecting the highest value here (8192 samples) - the system will still run comfortably.

System Clock

This shows the current clock state of the Nuendo 96/52 DSP system. The system is either Master (using its own clock) or Slave (AutoSync Ref).

Clock Modes - Synchronization

In the digital world, all devices are either the "Master" (clock source) or a "Slave" that synchronizes to the Master. Whenever several devices are linked within a system, there must always be a single master clock. The Nuendo 96/52 DSP's intelligent clock control is very user-friendly, being able to switch between clock modes automatically. Selecting "AutoSync" will activate this mode.

In AutoSync mode, the system constantly scans all digital inputs for a valid signal. If this signal corresponds to the current playback sample rate, the card switches from the internal quartz (AutoSync Ref displays 'Internal') to a clock generated from the input signal (AutoSync Ref displays "S/P-DIF" or "ADATx"). This allows on-the-fly recording, even during playback, without having to synchronize the card to the input signal first. It also allows immediate playback at any sample rate without having to reconfigure the card.
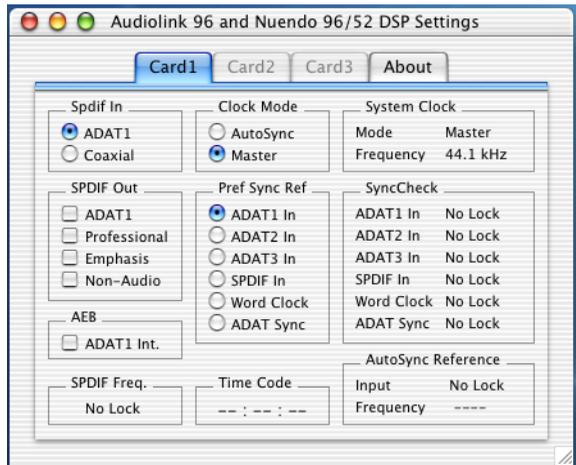
AutoSync guarantees that normal record and record-while-play will always work correctly. In certain cases however, e.g. when the inputs and outputs of a DAT machine are connected directly to the Nuendo 96/52 DSP, AutoSync may cause feedback in the digital carrier, so that synchronization breaks down. To prevent this, switch the Nuendo 96/52 DSP's clock mode over to "Master".

❒ **Remember that a digital system can only have one Master!**
If the Nuendo 96/52 DSP's clock mode is set to "Master", all other devices must be set to "Slave".

All the ADAT optical inputs in the Nuendo 96/52 DSP as well as the S/P-DIF input work simultaneously. However, as there is no input selector, the Nuendo 96/52 DSP must be informed which of the signals it should recognize as the sync reference (a digital device can only be clocked from a single source). This is why the system has been equipped with an automatic clock source selection, that adopts the first available input with a valid digital signal as the clock reference input. The input currently used as sync reference and its sample frequency are shown in the AutoSync Ref status field.

A preferred input can be defined via Pref. Sync Ref (Preferred Synchronization Reference). As long as the card recognizes a valid signal there, this input will be designated as the sync source. Otherwise the other inputs will be scanned in turn. If none of the inputs are receiving a valid signal, the card automatically switches clock mode to "Master".

Setting a "Pref Sync Ref" is essential. to cope with some situations which may arise during regular studio work. One example: An ADAT recorder is connected to the ADAT1 input (ADAT1 immediately becomes the AutoSync source) and a CD player is connected to the S/P-DIF input. Try recording a few samples from the CD and you will be disappointed. Few CD players can be synchronized. The samples will inevitably be corrupted, because the signal from the CD player is read with the (wrong) ADAT clock i.e. it is out of sync. In this case, "Pref Sync Ref" should therefore be temporarily set to S/P-DIF.

If several digital devices are to be used simultaneously in a system, they not only have to operate with the same sample frequency but also be synchronous with each other. This is why digital systems always need a single device defined as "Master", which sends the same clock signal to all the other ("Slave") devices. The SyncCheck technology lets you easily test and display the current clock status. The "SyncCheck" field indicates whether no signal ("No Lock"), a valid signal ("Lock") or a valid and synchronous signal ("Sync") is present at each of the digital clock source inputs. The "AutoSync Ref" display shows the current sync source and frequency.

In practice, SyncCheck gives you a very quick overview whether all digital devices connected to the system are configured properly. With SyncCheck, finally everyone can master this common source of error, previously one of the most complex issues in the digital studio world.

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┌─────────────────────────┐   ┌─────────────────────────────────┐
│        ADAT XT          │   │          Yamaha 02R             │
│ ┌────┐ ┌────┐ ┌────┐    │   │ ┌────┐ ┌────┐   ┌────┐ ┌────┐    │
│ │Sync│ │Opt.│ │Opt.│    │   │ │Opt.│ │Opt.│   │Opt.│ │Opt.│    │
│ │Out │ │ In │ │Out │    │   │ │ In │ │Out │   │ In │ │Out │    │
│ └────┘ └────┘ └────┘    │   │ └────┘ └────┘   └────┘ └────┘    │
└─────────────────────────┘   └─────────────────────────────────┘

┌──────────────────────────────────────────────────────────────┐
│ ┌────┐ ┌────┐ ┌────┐   ┌────┐ ┌────┐   ┌────┐ ┌────┐          │
│ │Sync│ │Opt.│ │Opt.│   │Opt.│ │Opt.│   │Opt.│ │Opt.│          │
│ │ In │ │Out │ │ In │   │Out │ │ In │   │Out │ │ In │          │
│ └────┘ └────┘ └────┘   └────┘ └────┘   └────┘ └────┘          │
│  Sync      ADAT3         ADAT2           ADAT1                 │
│                  Nuendo 96/52 DSP                              │
└──────────────────────────────────────────────────────────────┘
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An example to illustrate this: The ADAT1 and ADAT2 inputs receive signals from a digital mixing desk that has been set to clock mode "Internal" or "Master". An ADAT recorder is connected to the ADAT3 input. The Nuendo 96/52 DSP is set to AutoSync mode. As expected, SyncCheck shows that the ADAT1 and ADAT2 inputs are in sync (as they are driven by the same clock from the mixing desk), but shows "Lock" instead of "Sync" for the ADAT3 input. As the ADAT recorder does not receive any signals from the Nuendo 96/52 DSP or from the mixer, it generates its own clock at a rate that is (almost) the same as the sample frequency of the mixing desk – but not identical.

Solution: To drive the ADAT recorder from its digital input, set it to Slave mode (DIG) and connect the input to the Nuendo 96/52 DSP's ADAT3 output. The Nuendo 96/52 DSP is already in sync with the mixing desk, so it will send an identical (synchronous) signal to ADAT3 out. The ADAT recorder will lock onto this, its output will also be in sync. The signal from the ADAT recorder is now fully in sync with the signals from the mixing desk.

Thanks to its AutoSync technology and lightning fast PLLs, the Nuendo 96/52 DSP is not only capable of handling standard frequencies. It can also handle any sample rate between 25 and 105 kHz. The input selected in 'Pref Sync Ref' serves as the synchronization source. If Word clock is selected as input, this will serve as the synchronization source, offering any sample frequency between 25 kHz and 56 kHz in varispeed operation.

The current sample frequency at the S/P-DIF input (displayed in the 'S/P-DIF In' field) is useful for troubleshooting and controlling the configuration of all connected digital devices. If an input without a valid signal (or a faulty one) is selected, 'No Lock' will be displayed here. In varispeed mode, or if the sample frequency is way out of tune, 'Lock' is displayed.

At 88.2 or 96 kHz the sample frequency shown in the "S/P-DIF In" field differs from the one shown in "AutoSync Ref" if one of the ADAT inputs has been selected in "Pref Sync Ref". The card automatically switches to its Sample Split mode here, as ADAT optical inputs and outputs are only specified up to 48 kHz. Data from/to a single input/output is distributed onto two channels, the internal frequency remains at 44.1 or 48 kHz. In this case, the ADAT sample frequency is only half the S/P-DIF frequency.

Word Clock

Technical Description and Usage

Digital audio is subject to a basic clock, the sample frequency. The signal can only be processed or transferred correctly, if all devices in the system share the same clock. Otherwise digital signals are misinterpreted, which results in distortion, clicks/crackle and even drop-outs.

AES/EBU, S/P-DIF and ADAT are self-clocking. An additional line for Word clock could therefore be considered redundant. In practice however, using several devices at the same time *can* cause problems. For example, if devices are connected in a loop and no device has been defined as the "Master" device, self-clocking may break down. Besides, the clocks of all devices must be synchronized from a single source. Devices without S/P-DIF inputs (typically dedicated playback devices such as CD players) can not be synchronized via self-clocking.

In digital studios, synchronization requirements are met by connecting all devices to a central sync source. The master device could e.g. be a mixing desk that sends a reference signal – the Word clock – to all other devices. However, this will only work if all the other devices have Word clock inputs (some professional CD players do) which allows them to run as Slaves. If this is the case, all devices will receive the same clock signal and are in sync with each other.

Cables and Termination

Word clock signals are usually distributed in the form of a network, split with BNC T-adapters and terminated with resistors. We recommend using off-the-shelf BNC cables to connect all devices, as this type of cable is used for most computer networks. You will find all necessary components (T-adapters, terminators, cables) in most electronics and/or computer stores.

To avoid voltage loss and reflections, both the cable itself and the terminating resistor should have an impedance of 75 Ohm. If the voltage is too low, synchronization will fail. High frequency reflection effects can cause both jitter and sync failure.

Ideally, all outputs of Word clock delivering devices have a very low impedance, while all Word clock inputs have a high impedance, in order not to attenuate the signal when it passes through the chain. But there are also negative examples; e.g. when the 75 Ohms are built into the device and cannot be switched off. In this case the network load is often 2 x 75 Ohms and the user is forced to buy a special Word clock distributor. Note that such a device is generally recommended for larger studios. Furthermore, 75 Ohm cable is almost impossible to find these days. 50 Ohm cable is standard, but this will also work if the termination resistors are 75 Ohm.

The Nuendo 96/52 DSP's Word clock input can be high-impedance or terminated internally, ensuring maximum flexibility. If termination is necessary (e.g. because the Nuendo 96/52 DSP is the last device in the chain), you should set the jumper X105 on the Expansion Board into the 75 Ohm position.

If – on the other hand – the Nuendo 96/52 DSP sits within a chain of devices that receive Word clock, plug a T-adapter into its BNC input jack and the cable supplying the Word clock signal to one end of the adapter. Using another BNC cable, connect the free end to the next device in the chain. Remove jumper X105 from the Expansion Board. The last device in the chain should be terminated using another T-adapter and a 75 Ohm resistor (available as a short BNC plug). Devices with internal termination do not need T-adaptor and terminator plug.

General Operation

The green "Lock" LED on the Expansion Board will light up when the input receives a valid Word clock signal. Selecting "Word Clock" in the "Clock Mode" field will switch clock control over to the Word clock signal. As soon as a valid signal appears at the BNC jack, "AutoSync Ref" will display "Word". This message has the same function as the green "Lock" LED, but appears on the monitor, i.e. the user can check immediately whether a valid Word clock signal is present and is currently being used.

❐ **The Word clock output as well as all ADAT ports only work in Single Speed mode. At 96 kHz, the Word clock output will therefore be a 48 kHz signal.**

Using more than one Nuendo 96/52 DSP

The current drivers support multiple interfaces and any combination of I/O-boxes (Multiset/Digiset/Nuendo 96/52 DSP). Please note that (of course) only the ADAT Sync In of one system can be used. Additionally, all systems must work in sync, thus receive valid sync information (either via Word clock or AutoSync).

Special Characteristics of the S/P-DIF Output

Apart from the audio data itself, digital signals in S/P-DIF or AES/EBU format have a header containing the channel status information. A false channel status is a common cause of malfunction. The Nuendo 96/52 DSP ignores the received header and creates a totally new one for the output signal.

❑ **Note that in record or monitor modes, set emphasis bits will disappear. Recordings originally done with emphasis should therefore always be played back with the emphasis bit set!**

This can be done by selecting the "Emphasis" switch in the Settings dialogue ("S/P-DIF Out"). This setting is updated immediately, even during playback. The Nuendo 96/52 DSP's new output header is optimized for largest compatibility with other digital devices:

- 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz or 96 kHz, depending on the current sample rate
- Audio use, Non-Audio
- No Copyright, Copy Permitted
- Format Consumer or Professional
- Category General, Generation not indicated
- 2-channel, No Emphasis or 50/15 µs
- Aux bits Audio Use

Professional AES/EBU equipment can be connected to the Nuendo 96/52 DSP thanks to the transformer-balanced coaxial outputs, and the "Professional" format option with doubled output voltage. Output cables should have the same pinout as those used for input (see "), but with a male XLR plug instead of a female one.

❑ **Note that most consumer-orientated equipment (with optical or phono S/P-DIF inputs) will only accept signals in "Consumer" format!**

The audio bit in the header can also be set to "Non-Audio". This is necessary when Dolby AC-3 encoded data is sent to external decoders (surround-sound receivers, television sets etc. with AC-3 digital inputs), as these decoders would otherwise not recognize the data as AC-3.

Operation under ASIO 2.0

General

Start Nuendo and open the "Device Setup" dialog from the Devices menu.

Select "VST Multitrack" on the left side of the dialog and make the desired settings.

If necessary, read the corresponding sections in the Nuendo operation manual.

You can open the Nuendo 96/52 DSP Settings dialog from here by clicking onto the "Control Panel" button. (Windows version see , Macintosh- version see).
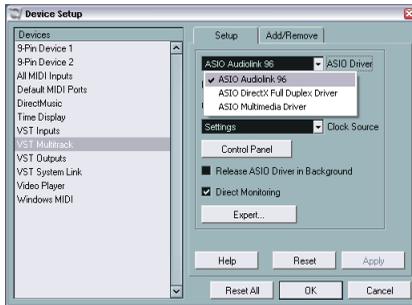
Under Windows, the Steinberg systems Nuendo 96/52 DSP and AudioLink use the same Settings dialog.

For this reason, the Settings dialogs and driver s of the Windows-and the Mac OS version have different names.
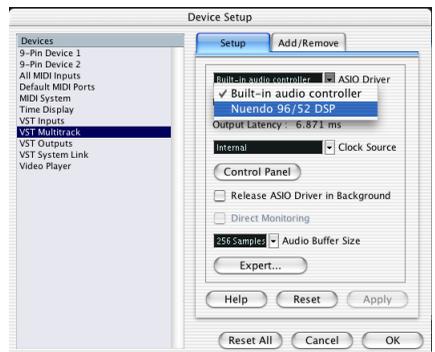
The Nuendo 96/52 DSP supports simultaneous recording and play-back of audio data in the S/P-DIF and ADAT formats. Please note that this only works if the external S/P-DIF devices run in sync. If they don't, recordings will be corrupted.

Furthermore the Nuendo 96/52 DSP supports "ASIO Direct Monitor-ing" (ADM).

Bei einer Samplefrequenz von 88,2 oder 96 kHz arbeiten der bzw. alle ADAT optical Ein- und Ausgänge im Sample Split Verfahren. Es ste-hen dann nur noch 4 (Multiset) bzw. 12 (Digiset) ASIO ADAT-Kanäle zur Verfügung.

If the sample frequency is set to 88.2 or 96 kHz, all ADAT optical in-puts and outputs operate in Sample Split mode. This reduces the number of available channels from 24 to 12.

Performance

Audio performance problems often originate in insufficient hard disk performance. If you record on multiple tracks and the first track is sud-denly missing or an "Audio: Record Error" message appears, then the disk sub-system is too slow, which means it is unable to write the au-dio data to the hard disk quickly enough. The problem can almost al-ways be solved by changing "Disk Block Buffer Size" from the default 64kB to 256kB.

This is especially true if you want to record more than 12 tracks at the same time. 26 tracks are only possible after changing "Disk Block Buffer Size" to 256kB (depending on which computer you use).

❐ **Currently, while this manual is written, changing Buffer Size is done dif-ferently under Windows and Mac OS. This will most probably change shortly. Currently, a Windows user can change the Buffer Size in the Nuendo 96/52 DSP Settings dialog and a Mac user can change it in the host application.**

The heyday of (expensive) SCSI hard disks in high-speed audio work-stations is over. Today's cheap high-capacity EIDE disks allow contin-uous transfer rates of well over 10 MB per second. In practical terms, this is more than enough to record up to 24 simultaneous tracks using Nuendo and Nuendo 96/52 DSP!

However, under Windows 98 the hard disks must use Busmaster drivers. To activate the EIDE Busmaster mode in Windows 9x, open the Device Manager (Control Panel/System). Double click on "Disk drives", then on the required hard disk(s). Select "DMA" in the Properties dialog, then restart Windows. Under Windows 2000/XP busmaster operation is default.

The Buffer Size value in Nuendo 96/52 DSP's Settings dialog or in the host application determines the latency (in this case the delay) between the audio application and the Nuendo 96/52 DSP as well as general system stability. The higher the value, the more tracks can be recorded and played back simultaneously and the longer the system takes to react. At the given maximum of about 0.2 seconds, you will not notice much delay at all - the system will still respond quickly and smoothly.

Most present systems are unable to use the 1.5 ms mode without audible clicks. Current PCs can handle 3 ms. For optimum reliability we recommend setting the highest latency possible, 182 ms.

Synchronization

To achieve sample-accuracy between the ADAT recorder and Nuendo 96/52 DSP while running Nuendo, connect the ADAT sync output to the 9-pin D-type sync input of the Nuendo 96/52 DSP. The "Time Code" field in the Settings dialog should now show the same position as the ADAT recorder.

To open the corresponding dialog, select "Sync Setup" in the Transport menu. Activate the "ASIO Positionioning Protocol" function by clicking on it. in the "Timecode Source section of the dialog. Now you can activate Sync mode by simply clicking on the Sync button in the Nuendo Transport Panel
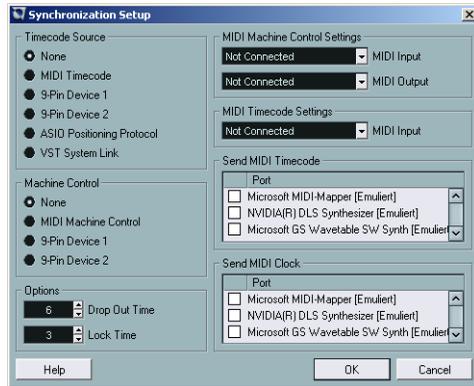
If synchronization doesn't work, i.e. Nuendo does not respond when the ADAT is set to "Play", please try the following:

- Check the cables.
- Switch Sync off and on again (in the Nuendo Transport Panel).
- Open the "Device Setup" dialog from the Devices menu and click on "Reset".
- Switch on the ADAT recorder(s) before starting Nuendo.
- Use the BRC as Master and send its Word clock to all other devices.
- Use the Clock Mode ADAT Sync.

Known Problems

If your computer doesn't have sufficient CPU power and/or sufficient PCI-bus transfer rates, then drop outs, crackling and noise will occur. To make sure that the problems are not caused by PlugIns that you might use, it is a good idea to deactivate them temporarily and try again.

Incorrect synchronization is another common source for trouble. ASIO does not support asynchronous operation, which means that the input and output signals must not only have the same sample frequency, but they must also be in sync. All devices connected to the Nuendo 96/52 DSP must therefore be properly configured for Full Duplex operation. As long as SyncCheck (in the Settings dialog) only displays "Lock" instead of "Sync", the devices have not been set up properly!

Under Windows, some newer UltraATA66 and UltraATA100 hard disk controller (also Raid controller) unfortunately seem to violate the PCI specifications. To achieve the highest throughput they hog the PCI bus, even in their default setting. Thus, when working with low latencies, heavy drop outs (clicks) become audible. Try to solve this problem by changing the default setting of the controller (for example by reducing the "PCI Bus Utilization").

Operation under GSIF (Gigasampler Interface)

Windows 98/SE/ME

The GSIF interface of the Nuendo 96/52 DSP's Windows 98/SE/ME driver allows direct operation with Gigasampler and Gigastudio, with up to 26 channels, 96kHz and 24bit. Additionally, the driver supports multi-client operation. For example ASIO can use channels 1/2 and Gigastudio (with GSIF) channels 3/4 simultaneously etc.

Gigasampler/Studio requires a lot of the computer's CPU power. An optimum performance is achieved by using a stand-alone GSIF PC. If this is not an option, we recommend that you set the ASIO latency to the highest value (186ms). This should give you a problem-free simultaneous operation of ASIO and GSIF. Gigastudio itself will still work at a very low latency.

- Please note: When used in multi-client operation, the Nuendo 96/52 DSP system requires identical formats! All programs simultaneously accessing the hardware *must* use the same sample rate.

❐ Simultaneous operation of GSIF and ASIO requires the use of different channels. As an example, Cubase e.g. always uses tracks 1/2 as its default setting, these tracks must therefore not be activated in Gigastudio/Sampler.

Windows 2000/XP

GSIF operation under Windows 2000/XP is basically similar to operation under Windows 9x, but it utilizes a modified interface that uses interrupts like ASIO. The user can set and change the latency (under Windows 9x, latency is fixed inside Gigastudio). When you use the Nuendo 96/52 DSP however, the latency will always be the same as the latency selected for ASIO operation. This can cause performance problems on slower machines when using GSIF and ASIO at the same time.

Please note that the Windows 2000/XP driver fully supports multi-client operation, including the combination MME/ASIO. So for example Cubase or Nuendo, Gigastudio and Sonar could be used simultaneously, provided each of these programs exclusively uses its own audio channels. Please also note that Gigastudio is running invisibly in the background, thus blocking its assigned audio channels, as soon as the Gigastudio MIDI ports are used – even when Gigastudio itself hasn't been started.

DSP Mixer: Routing and Monitoring

The Nuendo 96/52 DSP system includes a powerful digital real-time mixer. It more or less allows for unlimited simultaneous mixing and routing of all inputs and playback channels.

Here are some typical applications for DSP Mixer:

- Setting up delay-free submixes (headphone mixes).
- Unlimited routing of inputs and outputs (free utilisation, patchbay function).
- A signal can be distributed onto to several outputs at the same time.
- Simultaneous playback of different programs over only one stereo channel.
- Mixing of the input signal to the playback signal (complete ASIO Direct Monitoring).
- Integration of external devices (effects etc). in real-time.
- Mixdown of the three ADAT inputs into one (provides two additional inputs).

On you'll find a block diagram of the DSP Mixer mixer. It can help you understand the basic signal flow and routing. It shows that the record signal always remains unaltered, but can be passed on as often as desired, even with different levels. The level meter of inputs and playback channels are connected "pre-fader" (due to the enormous routing capabilities). The level meters of the hardware's outputs are connected "post-fader".

Elements on the User Interface

The visible design of the mixer is mainly determined by the architecture of the Nuendo 96/52 DSP system:

- Upper row: Hardware inputs. The level shown is that of the input signal, thus fader independent. Per fader and routing window, any input channel can be routed and mixed to any hardware output (third row).
- Middle row: Playback channels (playback tracks of the software). Per fader and routing window, any playback channel can be routed and mixed to any hardware output (third row).
- Lower row: Hardware outputs. Because they refer to the output of a subgroup, the level can only be attenuated here (in order to avoid overloads). Routing is not possible. This row has two additional channels, the analog outputs.

Every single channel has various elements:

Input and playback channels each have a mute and a solo button. Below each there is the panpot. In order to save space, this has been designed as an indicator bar (L/R).

In the section below the pan, the present level is displayed numerically as RMS or Peak. This is updated ca. every half a second. Overs are indicated here by an additional red dot.

Below this follow the fader and a level meter. The meter both shows peak values (zero attack, 1 sample is sufficient for full scale display) by means of a yellow line and mathematically correct RMS values by means of a green bar. The RMS display uses a relatively slow time constant, so that it shows the average loudness quite well.

The section below fader and level meter displays the current gain and panorama values.
The white area shows the channel name,
the black area shows the current routing target.

DSP Mixer – An Overview

The following sections explain all functions of the surface step by step. When you start the DSP Mixer, its last settings are recalled automatically. When you start it for the first time, a default file is used that connects all playback tracks 1:1 to the corresponding hardware outputs with a gain setting of 0 dB. The faders in the upper row are set to maximum attenuation (called m.a. in the following), so there is no monitoring of the input channels.
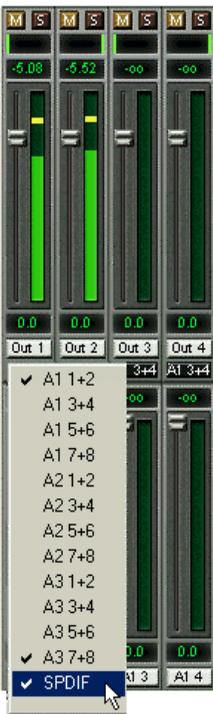
We will now create a small submix for the S/P-DIF output. Please start a multitrack playback and connect a D/A-converter to the S/P-DIF output. Click onto the routing area at the bottom of playback channel 1 ("Out 1"). A pop-up becomes visible, showing a checkmark in front of "A1 1+2". Select "S/P-DIF" here. The pop-up disappears and in the routing section, "A1 1+2" has been replaced by "S/P-DIF". Now move the fader with the mouse. As soon as the fader value is unequal m.a., the present state is stored and routing is activated. Move the fader button to ca. 0 dB. The present gain value is shown below the fader in green letters. In the lower row, on channels 25 and 26 (SP.L. and SP.R.), you can also see the level of what you are hearing. The level meter of the hardware output shows the outgoing level. Click into the panorama area above the fader and drag the mouse in order to set the panorama, in this case the routing between channels 25 and 26. The present pan value is also displayed below the fader.

Please carry out the same steps for "Out 2" now, in order to route it to the S/P-DIF output as well.

Often signals are stereo, i. e. a pair of two channels. It is therefore helpful to be able to make the routing settings for two channels at once. To do this, continuously hold down the [Ctrl]-key while you click into the routing section of "Out 3" and do the following. The routing list pops up with a checkmark at "A1 3+4". Select "S/P-DIF". Now, you can let go of the key and channel 4 has been set to "S/P-DIF" as well.

Depending on your mouse configuration, setting a fader to exactly 0 dB may be difficult. But the DSP Mixer offers a high resolution mode that "stretches" mouse movements by a factor of 8. Simply move the fader close to the 0 position, then press the [Shift]-key and move the fader handle to the exact position. In this mode, a gain setting accurate to 0.1 dB is no problem at all.

Please set "Out 4" to a gain of around -20 dB and the pan close to center. Now click onto the routing section. In the pop-up, you'll see two checkmarks, one next to "A1 3+4", the other one next to "S/P-DIF". Click onto "A3 7+8". The pop-up disappears, fader and panpot jump to their initial values and the signal can now be routed to the last ADAT channels. You can continue, until all entries have got a checkmark, i. e. you can send the signal to all outputs simultaneously.

You will certainly have noticed that the mix has not changed, while you were routing the channel to other outputs and setting different gain values. With all analog and most digital mixing desks, the fader setting would affect the level for every routed bus – not so for DSP Mixer. DSP Mixer allows for setting all fader values individually. Therefore the faders and the panpots jump to the appropriate setting as soon as another routing is selected.

You can delete the checkmarks and thus undo the routing by by moving the corresponding fader to its maximum attenuation position (all the way down). This setting deactivates the routing: why route if there is no level? Click onto "A1 3+4" in the routing pop-up, pull the fader down, open the pop-up again – the checkmark will be gone.

Submix View

Such a wide range of possibilities makes it difficult to maintain an overview. As described, practically all hardware outputs can be used for different submixes. And when opening the routing pop-up, you might see a multitude of checkmarks, but will not get an overwiev, i.e., how the signals come together and where. This problem is solved by using the "Submix" view mode . In this mode, all routing sections jump to and display any routing for the currently selected channel pair. So you can see immediately of which channels, which fader and pan settings a submix (for example "S/P-DIF") consists.

At the same time, the Submix View simplifies setting up the mixer, as all channels can be set to the same routing destination simultaneously with just one click.

Mute and Solo

Mute works pre-fader, thus mutes all active routings of the channel. As soon as any Mute button is pressed, the Master Mute button lights up in the quick access area. It can be used to switch all selected mutes on and off. You can use this to easily create mute groups that you activate and deactivate with this method.

The same applies to the Solo and the Master Solo buttons. Solo works as a solo-in-place. As soon as one Solo button is pressed, all other Mute buttons are activated and light up. But DSP Mixer would not be an Intelligent Audio Solution, if it didn't behave as you'd expect from a mixing console. If you, for instance, mute "Out 1" to "Out 4" and press Solo for "Out 5", of course all Mute buttons will light up. If you deactivate Solo, the Mute buttons for "Out 1" to "Out 4" will light up as before. And if you chose Solo for a channel of this Mute group, mute will be deactivated, but immediately activated again, as soon as Solo is released.

Hotkeys – DSP Mixer Key Combinations

DSP Mixer knows a few key combinations, that facilitate and speed up the mixer setup. The [Shift]-key for fader and panpot fine-tuning has already been mentioned. But the [Ctrl]-key can do far more than change the routing pairwise:

• Clicking anywhere into the fader area with the [Ctrl]-key pressed will set the fader to 0 dB, -6 dB for the hardware outputs.
• Clicking anywhere into the pan area with the [Ctrl]-key pressed will set the panorama to <C> meaning "Center" position.

The faders can also be moved pairwise, corresponding to the stereo-routing settings. This can be achieved by pressing the [Alt]-key and is especially comfortable when you set the S/P-DIF and analog output level. Even the Panorama settings can be operated with [Alt], from stereo through mono to inversed channels. But also the Mute and Solo buttons (ganged or inversed switching!). DSP Mixer also supports combinations of these keys. If you press [Ctrl] and [Alt] at the same time, clicking with the mouse makes the faders jump to 0 dB pairwise, and they can be set pairwise by [Shift]-[Alt] in high resolution mode.

Also very useful: the faders have two mouse sensitive areas. The first area is the fader button, which can be grabbed at any spot without changing its position. This avoids unwanted changes induced by just clicking somewhere on the button. The second area is the whole fader setting range. Clicking somewhere into this range lets the fader jump to the mouse position at once. For instance, if you want to set a fader to m.a. position, it is sufficient to click onto the lower end of the fader range. You can do this pairwise by simultaneously pressing the [Alt]-key.

Using the hotkeys [I], [O] and [P], the complete row each of Input, Playback and Output channels can be toggled between visible and invisible. The hotkey [S] lets you switch Submix view On or Off. These four hotkeys have the same functionality as the corresponding buttons in the View section of the Quick Access Panel. The Level Meter Setup dialog can be opened by pressing [L] (Macintosh) or [F2] (Windows PC).

Additional hotkeys are available to control the Level Meter settings (see):

Key [4] or [6]	Display range 40 or 60 dB
Key [E] or [R]	Numeric display shows Peak or RMS
Key [0] or [3]	RMS display is absolute or relative to 0 dBFS

The Quick Access Panel

This section includes additional options which further improve the DSP Mixer handling. The Master buttons for Mute and Solo have already been described. They facilitate group-based work.

In the View section, the single rows can be made visible or invisible. If you don't need the inputs because you work on a pure playback mix, then you can hide the complete upper row by clicking on the "Input" button. If you are currently also not interested in the hardware outputs, you can save even more space and reduce the view to the playback channels only. All combinations are possible.

"Submix" sets all routing windows to the same selection as described before. Deactivating "Submix" automatically recalls the previous view.

You can also make the Mixer window smaller horizontally und scroll the currently visible part. If you only want to monitor or set just a few channels or level meters, you can substantially reduce the DSP Mixer size and thus save space on the screen.

The Presets are one of the most powerful and useful features of the DSP Mixer. The eight buttons represent eight Preset files (see next section). These each contain a complete mixer setting. You can change the settings of all faders and other parameters in real-time with a single mouse click on one of the Preset buttons. The Save button lets you store the current settings in the currently selected Preset memory. Presets thus let you easily change back and forth between a signal distribution, complete input monitoring, a stereo and mono mix and various submixes.

If you change a parameter in the currently active Preset (e. g. by moving a fader), the Preset button starts to flash to remind you that something was changed. At the same time, this reminds you on which original Preset your current setting is based.

If you load a Preset using "Open file" on the File menu, no Preset button will be illuminated. Mixer settings can of course be saved as usual and using long file names.

Up to three Nuendo 96/52 DSP systems can be used simultaneously. The Card buttons switch between the systems. "Systems", because card 1 can be a Digiset, card 2 could also be a Multiset etc.

The number of ADAT channels is reduced to half the number automatically when you select double speed operation (88.2 or 96 kHz). The display is adjusted accordingly and all fader settings remain stored (even the invisible ones).

Presets

During the driver installation 8 factory Presets will be copied to the Windows directory (Windows) or the Preferences folder (Macintosh). The files are named preset1.mix to preset8.mix. These files are used when you click on one of the 8 Preset buttons in the Quick Access Panel.

The DSP Mixer only reads these files when used for the first time. As soon as any of the Presets is saved anew, the DSP Mixer writes a new file and adds the number of the currently used system (Card 1, 2 or 3). The file "preset1.mix" thus changes to "preset11.mix" if Card 1 was active. This method offers two major advantages:

• Presets modified by the user will not be overwritten when you reinstall or update the driver software.
• The factory Presets remain unchanged and can be reloaded anytime using the Open function on the File menu.

The eight factory Presets do not only offer useful settings for the DSP Mixer, but also represent a pretty good base for modification that meets your personal requirements.

Preset1.mix

Description: All channels routed 1:1, playback monitoring via headphone out.

Details: All inputs are set to maximum attenuation (m.a.). All playback channels to 0 dB and routed to the same output. All output channels are set to 0 dB, Phones to -6 dB. Submix of all inputs and outputs to the analog output (Phones), with input faders set to m.a., playback to 0 dB. All channels prepared for all routings to left/right panning. Level display is set to RMS -3 dB.

• **Note: This preset is Default. It offers the standard functionality of an I/O-card.**

Preset2.mix

Description: All channels routed 1:1, input and playback monitoring via Phones. Rest same as Preset 1, plus submix of all inputs (0 dB) on Phones.

Preset3.mix

Description: All channels 1:1, input and playback monitoring via Phones and outputs. As Preset 2, but all inputs set to 0 dB (1:1 pass through).

Preset4.mix

Description: All channels 1:1, playback monitoring via Phones and outputs. As Preset 3, but all inputs muted.

Preset5.mix

Description: All faders m.a. As Preset 1, but all outputs m.a.

Preset6.mix

Description: Submix on S/P-DIF at -6 dB. As Preset 1, plus submix of all playbacks on S/P-DIF. View Submix S/P-DIF is active.

Preset7.mix

Description: Submix on S/P-DIF at -6 dB. As Preset 6, but submix of all inputs and outputs on S/P-DIF. View Submix S/P-DIF is active.

Preset8.mix

Description: Panic. As Preset 4, but also playback muted (no output signal)

Monitor – A Function of the Windows MME Driver

The Monitor section of the Quick Access Panel is only applicable if you use the Windows MME driver, i.e. when using programs like WaveLab, Soundforge, Sonar or Samplitude.

Monitor offers two advanced automated monitoring solutions. Monitoring will either be controlled by special commands directly from the recording software (Samplitude/Sequoia/SAWStudio, ZLM-mode) or by the recording status itself (Mix/Replace mode).

❐ **The basic method used is as simple as it is clever: ZLM and record control the Mute buttons of the corresponding channels. For this to work, Mute must be activated on the Input channel that you intend to record on. The fader must not be set to m.a..**

A click on Load Def. will load a template (preset4.mix), which can be used to verify and test this functionality. But it is also possible to use any other mixer state, for as long as the recording channels are muted manually.

As soon as a recording starts, the corresponding channels will be unmuted, i.e. the input signal will be processed according to the current mixer settings. DSP Mixer lets you check the whole process visually, as the Mute buttons in the DSP Mixer will be switched on and off automatically.

In Mix mode, the input signal will be mixed onto the outputs when record is active. In Replace mode, the Mute button of the corresponding playback channel will be activated, so that the input signal replaces the playback signal.

ZLM is a special function for tape machine style monitoring when you use punch-ins and outs. For this to work, the option "Hardware monitoring during punch" must be activated in Samplitude/Sequoia. Then each punch-in will deactivate the corresponding Mute buttons and each punch-out reactivates them.

All settings can be changed and set in real-time.

Level Meter

All calculations of the Peak, RMS and Over values are done in hardware. This makes the card independent of the software you use and significantly reduces CPU load.

The level meters integrated in the DSP Mixer include many useful functions.

Peak and RMS are displayed for each channel. You can select the different options by opening "Level Meter Setup" from the Options menu or by pressing the corresponding hotkey on the keyboard:
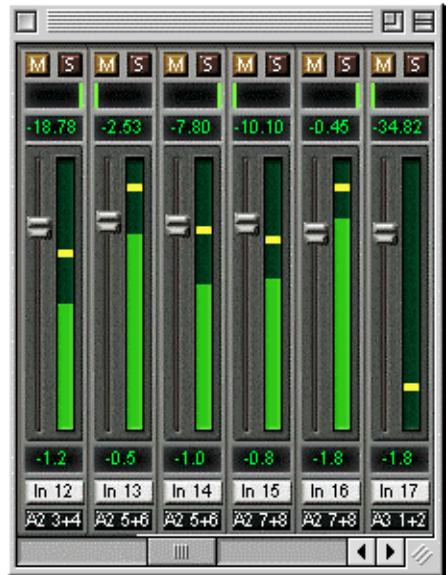
- Display range 40 or 60 dB (hotkey [4] or [6])
- Release time of the Peak display (Fast/Medium/Slow)
- Switch the Numeric display to Peak or RMS (Hotkey [E] or [R])
- Set the number of consecutive samples for Overload display (1 to 15)
- RMS display absolute or relative to 0 dBFS (Hotkey [3] or [0])

The last point ist often overlooked, but still important. RMS shows 3 dB less for sine signals. This is mathematically correct, but not very useful for a level meter. Here, a full scale sine signal thus shows both 0 dBFS Peak and RMS. This setting also results in directly readable signal-to-noise values.

The value displayed in the text field is independent of the setting 40/
60 dB. It represents the full 24 bit range of the RMS measurement.
This in turn allows for a SNR measurement "RMS unweighted" which
would otherwise only possible by using extremely expensive measure-
ment equipment.

This level display will constantly remind you of the reduced dynamic
range of your equipment, or even the whole studio. It's nice to have
everything running at 24 bit - and still there's noise and hum every-
where in the range around -90 dB… Sorry, but this is the hard reality.
The up-side is that the DSP Mixer lets you constantly monitor the sig-
nal quality without additional effort. It can thus be a very valuable tool
for sound optimization and error removal in the studio.

❐ **T measure the SNR (Signal to Noise), press [R] (for RMS) and [0] (refer-
ring to 0 dBFS, a full scale signal). The text field will then show similar
values as an expensive professional measurement system in "RMS un-
weighted" mode.**

• **Note: There is no RMS calculation for the third row, the physical outputs.
Therefore these green bars show the peak value.**

Problems & Solutions

General

The latest information can always be found on our website: www.steinberg.net

The ADAT timecode is not in sync

- The tape is formatted to 48 kHz, but played back at 44.1 kHz (Pitch). This "Blackface" problem cannot be solved in a satisfactory way.

ADAT timecode is running, but Cubase does not start "Play" automatically

- The input displayed in "Sync Ref" is not in sync mode. Sync mode is essential, because ADAT's so-called time code is really a sample position and is therefore only valid for synchronous audio data.
- Sync is displayed (referring to the card's clock), but the incoming data is not in sync with the sample position received at the ADAT Sync In. Then Cubase does not start. Solution: Set "Pref. Sync Ref" to the input corresponding to the received ADAT Sync signal.
- Sync mode wasn't activated (button in the Transport Panel), or ASIO 2.0 has not been chosen as the SMPTE sync source.

The input signal can not be monitored in real-time

- ASIO Direct Monitoring has not been enabled, and/or monitoring has been disabled globally.

The first 8 channels don't seem to work

- S/P-DIF output has been switched to ADAT1. This means that the first ADAT output device, and therefore the first 8 channels in the ASIO application, are no longer available. All channels and their assignments still exist, but the optical transmitter has been disconnected from the ADAT and is now fed from the S/P-DIF output (channels 25 and 26).

Playback works, but record doesn't:

- Check that there is a valid signal at the input. If so, the current sample frequency is displayed in the Settings dialog.
- Check whether the Nuendo 96/52 DSP has been selected as recording device in the audio application.
- Check whether the sample frequency set in the audio application ("Recording properties" or similar) matches the input signal.
- Check that cables/devices have not been connected in a closed loop. If so, set the systems's clock mode to "Master".

Crackle during record or playback:

- Increase the number and size of buffers in the "Settings" dialog or in the application.
- Try different cables (coaxial or optical) to rule out any defects here.
- Check that cables/devices have not been connected in a closed loop. If so, set the system's clock mode to "Master".
- Increase the buffer size of the hard disk cache.

PC only

- Activate Busmaster mode for the hard disks (see section "Performance" on page 44).
- In case of a recently done BIOS update of the motherboard: Propably "Load BIOS Defaults" was loaded instead of "Load Setup Defaults". This sets the "PCI Latency Timer" to 0 (default: 32).

Low Latency ASIO operation under Windows 2000/XP on single CPU systems:

- To use ASIO at lowest latencies under Windows 2000/XP even when only having one CPU, the system performance must be optimized for background tasks. Go to Control Panel/System/Advanced/Performance Options. Change the default "Applications" to "Background tasks". The lowest usable latency will drop from 23 ms to around 3 ms. This is no issue when using dual CPU systems.

Installation – Windows PC

After installation, the Nuendo 96/52 DSP can be found in the Device Manager (Settings/Control Panel/System, category "Sound-, Video- and Gamecontroller"). A double click on "Nuendo 96/52 DSP" starts the properties dialog. Choosing "Resources" shows Interrupt and Memory Range.

The latest information on hardware problems can always be found on the Steinberg website: www.steinberg.net

The dialog "New hardware component found" does not appear:

• Check whether the PCI interface is correctly inserted in the PCI slot.

The card and drivers have been installed correctly, but playback does not work:

• Check whether the Nuendo 96/52 DSP appears in the Device Manager. If the "Nuendo 96/52 DSP" device has a yellow exclamation mark, then there is an address or interrupt conflict.
• Even if there is no yellow exclamation mark, it is worth checking the "Resources" tab anyway.
• Check whether the Nuendo 96/52 DSP has been selected as current ASIO device.

The computer crashes whenever the Nuendo 96/52 DSP is accessed:

• If your graphics board is an older Matrox Mystique or uses a "968" S3 chip, there could be a memory allocation error. Change the memory area allocated to the Nuendo 96/52 DSP via /Resources/Change Setting.

Installation – Macintosh Computer

The dialog "New hardware component found" does not appear:

• Check whether the PCI interface is correctly inserted in the PCI slot.

The card and drivers have been installed correctly, but playback does not work:

• Check whether the Nuendo 96/52 DSP has been selected as current ASIO device.

The following symptoms are typical for PCI related problems:

• When booting the control panels are displayed too big, or cluttered across the whole screen
• Software or OS crash as soon as the card is used
These problems were reported with older computers (prior to G3). They can be solved in most cases by simply using a different slot, or by exchanging slots with other PCI cards (like SCSI controllers or graphics cards).

Software and Hardware Compatibility – Macintosh Computer

Nuendo 96/52 DSP is fully compatible with PCI bus version 2.1.

The Nuendo 96/52 DSP is compatible to all major ASIO applications, like Nuendo, Cubase SX/SL, Cubase VST, emagic Logic, Opcode Studio Vision PRO, Prosoniq SonicWORX, TC SPARK, Peak from Bias, Motu Digital Performer, Max/MSP from Cycling '74 and Super Collider.

As far as we are aware, the Nuendo 96/52 DSP digital inputs and outputs are fully compatible with all devices with S/P-DIF or AES/EBU interfaces.

Warranty

Each individual Nuendo 96/52 DSP undergoes comprehensive quality control and a complete test in a PC environment before shipping. This may cause very slight signs of wear (if it looks like it was used before – it was). We use high grade components which allows us to offer a full two year warranty. We accept a copy of the sales receipt as valid warranty legitimation.

Steinberg's replacement service within this period is handled by the retailer. If you suspect that your card is faulty, please contact your local retailer. The warranty does not cover damage caused by improper installation or maltreatment. In such cases replacement or repair can only be carried out at the owner's expense.

Steinberg does not accept claims for damages of any kind, especially consequential damage. Liability is limited to the value of the Nuendo 96/52 DSP. The general terms of business of the Steinberg Media Technologies GmbH apply at all times.

Appendix

Steinberg news, driver updates and other product information is available on our website:

http://www.steinberg.net

Trademarks

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Manual by Matthias Carstens

Editing of the Steinberg manual version: d.popow@musicandtext.com

Current driver version when this was written: Macintosh: 2.10, Windows 98: 2.4, Windows 2000/Windows XP: 2.51

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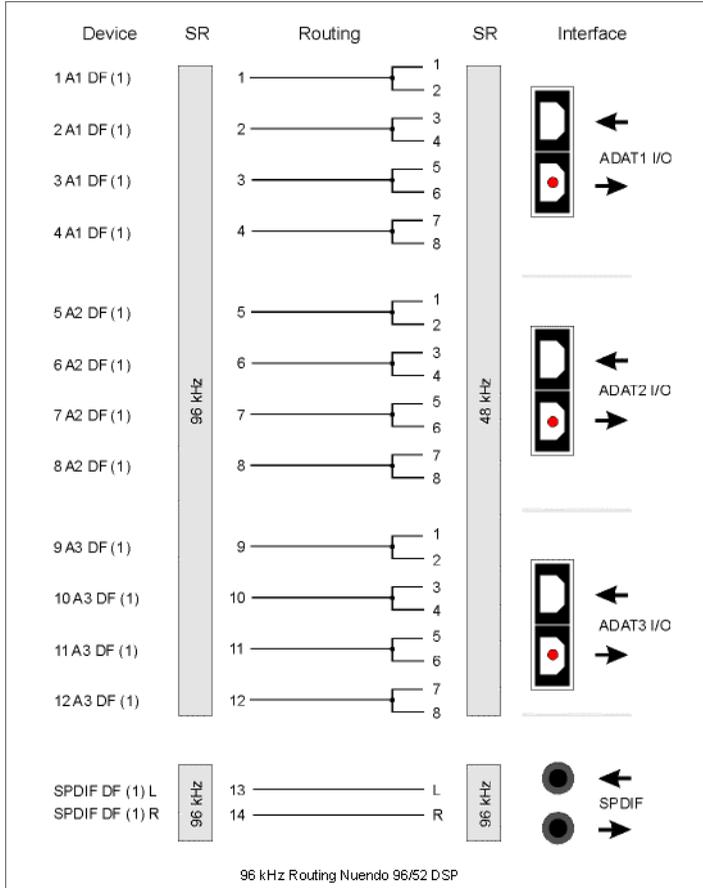
Diagrams

Block Diagram Nuendo 96/52 DSP
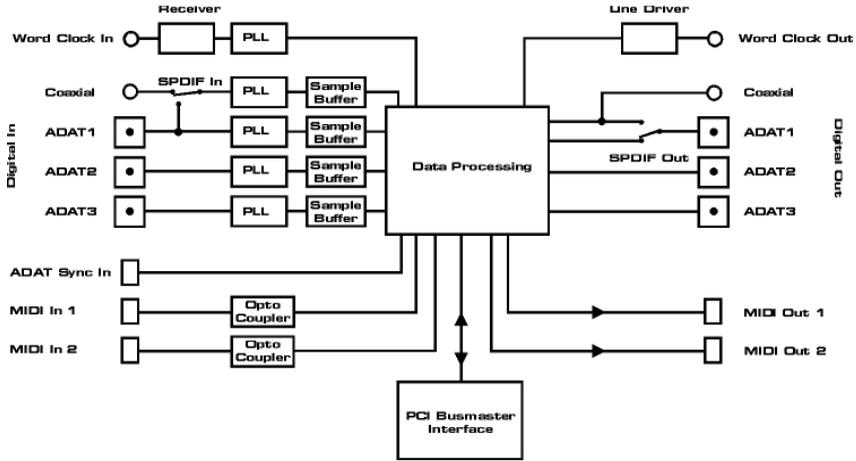
ADAT Track Routing, ASIO at 96 kHz

This diagram shows the signal paths in ASIO double speed mode (88.2 / 96 kHz). The devices available under ASIO have been implemented according to the hardware. Signal routing is identical for record and playback.


96 kHz Routing Nuendo 96/52 DSP

Device: The device name in the audio application, SR: Sample Rate

Device name code: Channel in ASIO host, ADAT interface, Digiface, card number

ADAT Track Routing, MME at 96 kHz

This diagram shows the signal paths in MME double speed mode (88.2 / 96 kHz). The devices available via wave driver have been designed to avoid conflicts in normal operation, which is why channels 5, 6, 7 and 8 of each ADAT device have been omitted. Signal routing is identical for record and playback.
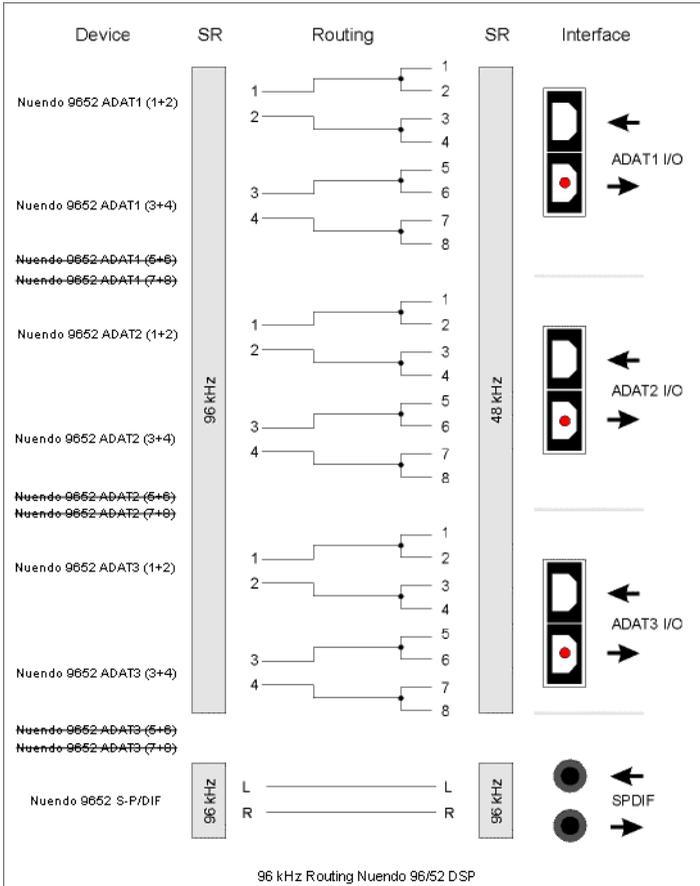

Device: The device name in the audio application, SR: Sample Rate

Block diagram DSP Mixer

This diagram shows the signal flow inside the DSP Mixer mixer of the Digiface. It shall clarify the following function:

• The input signal of the hardware (ADAT/S/P-DIF In) is always directly fed through to the recording software. At the same time it can be routed to all 26 hardware outputs (ADAT/ S/P-DIF), even to all of them simultaneously.
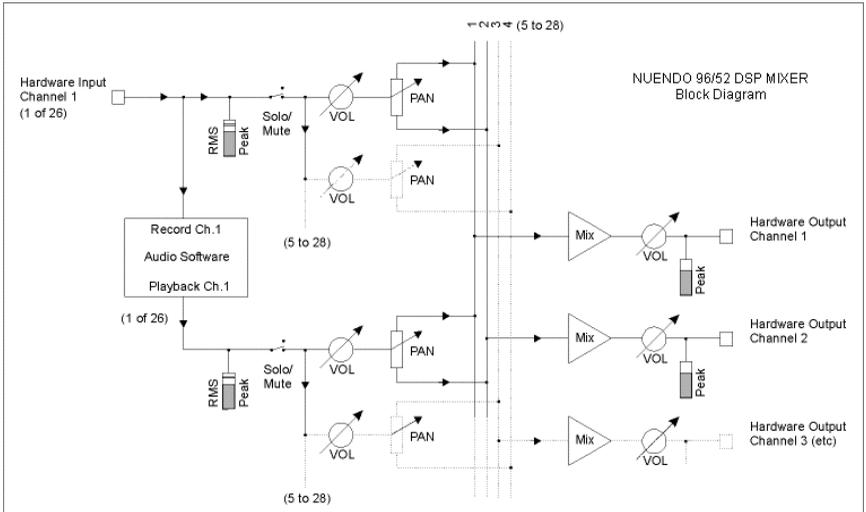
CE and FCC Compliance Statements

CE

This device has been tested and found to comply with the EN55022 class B and EN50082-1 norms for digital devices, according to the European Council directive on counterpart laws in the member states relating to electromagnetic compatibility (EMVG).

FCC

This device has been tested and found to comply with the requirements listed in FCC Regulations, part 15 for Class "B" digital devices. Compliance with these requirements provides a reasonable level of assurance that your use of this product in a residential environment will not result in harmful interference with other electronic devices.

This equipment generates radio frequencies and, if not installed and used according to the instructions in the User's Guide may cause interference harmful to the operation of other electronic devices.

Compliance with FCC regulations does not guarantee that interference will not occur in all installations. If this product is found to be the source of interference, which can be determined by turning the unit off and on again, please try to eliminate the problem by using one of the following measures:

• Relocate either this product or the device that is being affected by the interference
• Use power outlets on different branch circuits, or install AC line filters
• Contact your local retailer or any qualified radio and television engineer

When connecting external devices to this product, compliance to limits for a Class "B" device requires the use of shielded cables.

FCC compliance statement: Tested to comply with FCC standards for home or office use.