

User's Guide



Project Hammerfall[®]

DIGI 9636

DVD
24 Bit / 96 kHz ✓ *ready*



SyncAlign[®]

ZLM[®]

SyncCheck[®]



PCI Busmaster Digital I/O Card
2 + 16 Channels Stereo / ADAT Interface
24 Bit / 96 kHz Digital Audio
ADAT Sync In

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1. Introduction

Thank you for choosing the DIGI9636. This card is capable of transferring digital audio data directly to a computer from practically any device equipped with a digital audio interface, be it SPDIF, AES/EBU or ADAT optical. Installation is simple, even for the inexperienced user, thanks to the latest Plug and Play technology and full interrupt-sharing. The numerous unique features and well thought-out configuration dialogue puts the DIGI9636 at the very top of the range of digital audio interface cards.

The package includes drivers for Windows (98/NT/2000/XP) and MacOS. An ALSA driver for Linux or Unix is also available (see chapter 7.4).

Our high-performance philosophy guarantees maximum system performance by executing all functions directly in hardware and not in the driver (i.e. the CPU).

The optional Expansion Board turns the DIGI9636 into the fully equipped DIGI9652 (Project Hammerfall). After purchase of the Expansion Board please use the manual of the DIGI9652.

2. Package Contents

Please check that your DIGI9636 package contains each of the following:

- DIGI9636 PCI card
- Quick Info guide
- RME Driver CD
- Adapter cable (D-type to D-type/phono)
- Internal cable (2-core)
- 2 optical cable (TOSLINK), 2 m (6.6 ft)

3. System Requirements

- Windows 98/NT/2000/XP, MacOS or Linux
- A free PCI rev. 2.1 Busmaster slot

4. Brief Description and Characteristics

- PCI Busmaster interface with additional burst FIFO
- Hammerfall design: 0% (zero!) CPU load, even using all 36 ASIO channels
- All settings can be changed in real-time
- Enhanced mixed mode: ADAT In, SPDIF 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 8 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
- Enhanced ZLM prevents noises during asynchronous full duplex transfer
- SyncAlign guarantees sample aligned and never swapping channels
- SyncCheck tests and reports the synchronization status of input signals
- Full interrupt-sharing

5. Technical Specifications

5.1 Digital

- Ultra-low jitter S/PDIF: < 1 ns in PLL mode (44.1 kHz, optical in)
- Ultra-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 V_{ss} 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 (wordclock) supported

5.2 Digital Interface

- Phono input and output ground-free transformer coupled
- Connectors: optical (TOSLINK), phono, internal (CD-ROM/Sync in, Sync out)
- Clocks: ADAT Sync In, word clock I/O
- Formats: SPDIF (Consumer and Professional), ADAT optical

5.3 Transfer Modes: Resolution / Bits per Sample

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. The card works internally with 32-bit data, but audio data transfer is limited to 24-bits. As no additional Channel Status bits are transferred RME's TMS (Track Marker Support) is not available. Furthermore, DIGICheck's *Channel Status Display* is deactivated.

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)

Channel Interleave operation is not supported. As no additional Channel Status bits are transferred RME's TMS (Track Marker Support) is not available. Furthermore, DIGICheck's *Channel Status Display* is deactivated.

6. Hardware Installation



Before installing the DIGI9636, 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. Remove 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 DIGI9636 from its protective bag, discharge any static in your body by touching the metal chassis of the PC.
4. Insert the DIGI9636 firmly into a free PCI slot, press and fasten the screw.
5. Replace the computer's housing.
6. Reconnect all cables including the power cord.

7. Driver Installation

7.1 Windows 98/SE/ME

After the hardware has been installed correctly (see 6. Hardware Installation), and the computer has been switched on, Windows will recognize the new hardware component and start its 'Add New Hardware Wizard'. Insert the RME Driver CD into your CD-ROM drive, and follow further instructions which appear on your computer screen. The driver files are located in the directory **\\Hammerfall W98** on the RME Driver CD.

Windows will install the DIGI9636 driver, and will register the card in the system as a new audio device. The computer should now be re-booted.



Unfortunately, there are rare cases where the CD-ROM path (i.e. its drive-letter) has to be typed in again during the copy process.

The Hammerfall is configured using the DIGI9636 driver's Settings dialog (see section 9.1)

7.2 Windows NT

As automatic hardware recognition has not been implemented in Windows NT 4.0 the drivers have to be installed 'by hand'.

After the hardware has been installed correctly (see 6. Hardware Installation) and Windows NT has been booted, insert the RME Driver CD into your CD-ROM drive. Register the new device by starting >Control Panel /Multimedia /Devices /Audio Devices /Add<. Change the directory to **\\nt** in the CD-ROM. Windows NT will now install the driver. The RME Settings dialog will open automatically.

A click on 'OK' finishes the installation. After a reboot the DIGI symbol will show up in the systray of the taskbar. The DIGITray tool will be loaded automatically each time when booting NT.

A left mouse click on the DIGI96 symbol starts the 'Settings' dialog. The NT driver supports any combination of up to three RME cards. The driver is installed only once for all cards in the system.

7.3 Windows 2000/XP

After the hardware has been installed correctly (see 6. Hardware Installation), and the computer has been switched on, Windows will recognize the new hardware component and start its 'Hardware Wizard'. Insert the RME Driver CD into your CD-ROM drive, and follow further instructions which appear on your computer screen. The driver files are located in the directory **Hammerfall W2k** on the RME Driver CD.

Windows will install the DIGI9636 driver, and will register the card in the system as a new audio device. The computer should now be re-booted.

The Hammerfall is configured using the DIGI9636 driver's Settings dialog (see section 9.1)

7.4 Deinstalling the Drivers

A deinstallation of the Hammerfall's driver files is not necessary – and not supported by Windows anyway. Thanks to full Plug & Play support, the driver files will not be loaded after the hardware has been removed. If desired these files can then be deleted manually.

Unfortunately Windows Plug & Play methods do not cover the additional autorun entries of the Settings dialog and the registering of the ASIO driver. Those entries can be removed from the registry through a software deinstallation request. This request can be found (like all deinstallation entries) in *Control Panel, Software*. Click on the entry 'RME Hammerfall Tray Tools', or 'RME Hammerfall'.

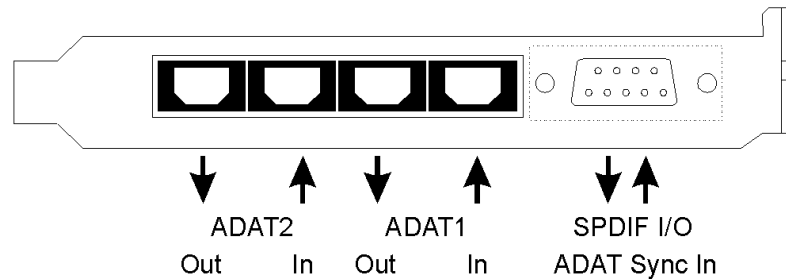
7.5 Linux/Unix

An ALSA driver for Linux/Unix and further information on ALSA is available at <http://www.alsa-project.org>

8. Operation and Usage

8.1 External Connections

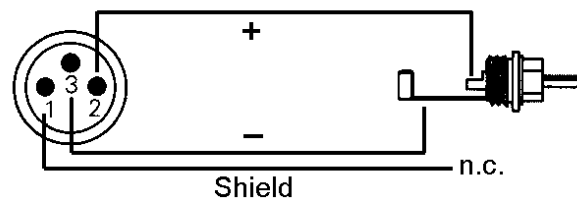
The card's bracket has two ADAT optical inputs and two ADAT optical outputs, as well as a 9-pin D-type socket. Coaxial SPDIF input and output requires plugging in the adapter cable, whereby the red phono socket is the output. The ADAT1 I/O next to the D-type socket can also be used for optical SPDIF, if this mode is selected in the Settings dialog.



An input is selected via the Settings dialog (started by clicking on the hammer symbol in the system tray). Hammerfall accepts the commonly used digital audio formats, SPDIF as well as AES/EBU. Channel status and copy protection are ignored.

In SPDIF mode, identical signals are available at both the optical and the coaxial outputs. An obvious use for this would be simply connecting two devices, i.e. using the DIGI9636 as a splitter.

To receive signals in AES/EBU format, an adapter cable is required. Pins 2 and 3 of a female XLR plug are connected individually to the two pins of a phono plug. The cable shielding is only 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.

8.2 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 a direct transfer of digital audio data within the computer.

Since board revision 1.5 it also allows to use an AEBx-I, when a 3-wire cable connects ST7 on both AEBx-I and Hammerfall.

Additionally board revision 1.5/1.6 offers two internal outputs, labeled ADAT1OUT (ST5) and ADAT2OUT (ST9). These can be used to operate one AEBx-O each, for a maximum of 16 analog outputs. The audio data are the same as on the corresponding optical output.

8.3 Windows MME Playback

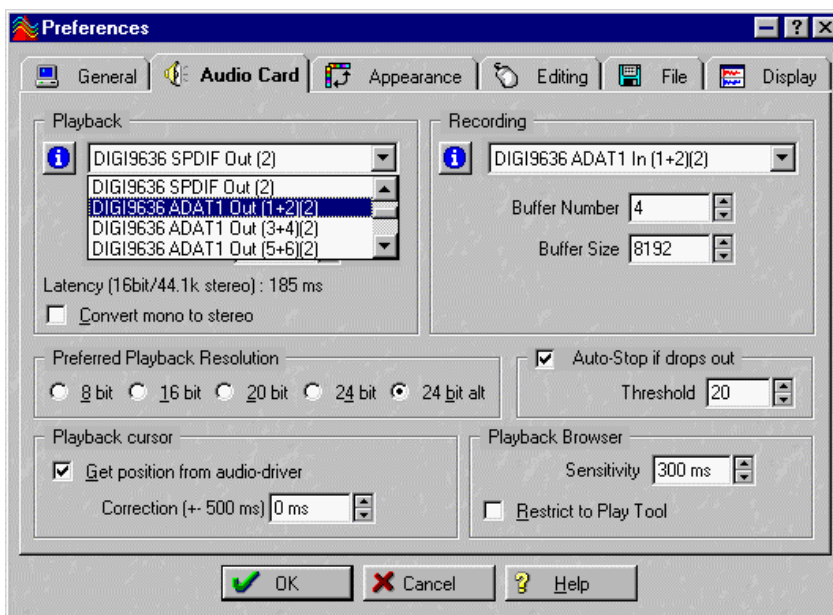
The Hammerfall can play back audio data only in supported formats (sample rate, bit resolution). Otherwise an error message appears (for example at 22 kHz and 8 bit).

In the audio application being used, DIGI9636 must be selected as output device. This can often be found in the *Options*, *Preferences* or *Settings* menus under *Playback Device*, *Audio Devices*, *Audio* etc. We recommend using 24-bit resolution for playback, to make full use of the DIGI9636's potential.

We strongly recommend switching all system sounds off (via >Control Panel /Sounds<). Also Hammerfall should not be the *Preferred Device* for playback, as this could cause loss of synchronization and unwanted noises. If you feel you cannot do without system sounds, you should consider buying a cheap Blaster clone and select this as *Preferred Device* in >Control Panel /Multimedia /Audio<.

The RME Driver CD includes step by step instructions for configuring many popular audio applications, found in the directory `\rmeaudio.web\english\techinfo\conf`.

The screenshot to the right shows a typical configuration dialogue as displayed by a (stereo) wave editor. After selecting a device, audio data is sent either to S/PDIF 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 will not arise because the DIGI9636 always reports the current play position correctly (even while recording - essential for chase lock synchronization).

The DIGI9636 ADAT optical interface allows sample rates of up to 96 kHz using a standard ADAT recorder. Single-channel data at this frequency requires two ADAT channels, achieved using the 'Sample Split' technique. This reduces the number of available ADAT channels from 16 to 8. Under Windows MME, channels are routed to ADAT devices in double-speed mode as follows:

- Only 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', section 21. Routing for record and playback is identical.

8.4 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, RME has included two unique features in the DIGI9636: a comprehensive I/O signal status display (showing sample frequency, lock and sync status) in the Settings dialogue, and the protective *Check Input* function.

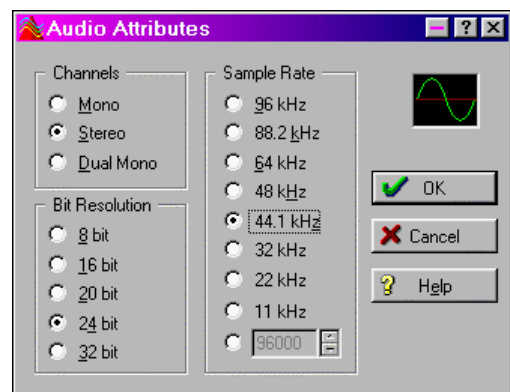
If a 48 kHz signal is fed to the input and the application is set to 44.1 kHz, *Check Input* stops the system from recording. This prevents faulty takes, which often go unnoticed until later on in the production. Such tracks appear to have the wrong playback rate - the audio quality as such is not affected.

The sample frequency shown in the Settings dialog (see chapter 9, screenshot Settings) is useful as a quick display of the current configuration (the board itself and all connected external equipment). If no sample frequency is recognized, it will read 'No Lock'.

With this configuring any suitable audio application for digital recording is simple. After selecting the required input, DIGI9636 displays the current sample frequency. This parameter can then be changed in the application's audio attributes (or similar) dialogue.

The screenshot to the right shows a typical dialogue 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 about 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 space on the hard disk.



It often makes sense to monitor the input signal or send it directly to the output. The DIGI9636 includes a useful input monitor function for just this purpose (see *Monitoring* in the Settings dialog). Activating Record or Pause in the application causes the input signal to be routed directly to the corresponding output. However, some applications block monitoring by constantly activating playback, even if the track is empty. This is often required by programs to ensure that timing and punch I/O will work correctly.

Currently two solutions exist which enable real-time monitoring even when when playback is active. Our ZLM (Zero Latency Monitoring) technology allows monitoring in Punch I/O mode - with this the card 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'.

The other solution is Steinberg's ASIO protocol with our ASIO 2.0 drivers and all ASIO 2.0 compatible programs. When 'ASIO Direct Monitoring' has been switched on the input signal is routed in real-time to the output whenever Record is started.

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

When using popular DVD software player like WinDVD and PowerDVD, their audio data stream can be send to any AC-3/DTS capable receiver, using the Hammerfall's SPDIF output. For this to work the SPDIF output wave device has to be selected in 'Control Panel/Sounds and Multimedia/Audio'. Also check 'use preferred device only'.

You will notice that the DVD software's audio properties now allow to use 'SPDIF Out' or to 'activate SPDIF output'. When selecting these, the software will transfer the non-decoded digital multichannel data stream using the RME card.

This 'SPDIF' signal sounds like chopped noise at highest level. Therefore check 'Non-audio' in the card's Settings dialog, to prevent most SPDIF receivers from accepting the signal, and to prevent any attached equipment from being damaged.

Setting the card to be used as system playback device is against common sense, as professional cards are not specialized to play back system sounds, and shouldn't be disturbed by system events. To prevent this, be sure to re-assign this setting after usage, or to disable any system sounds (tab Sounds, scheme 'No audio').

Note: The DVD player will be synced backwards from the RME card. This means when using AutoSync and/or word clock, the playback speed and pitch follows the incoming clock signal.

8.6 Low Latency under MME (Buffer Size Adjustment)

Using Windows 95 or 98 the MME buffer size was nothing to worry about. Latencies below 46 ms were not possible. Meanwhile both computers and operating system have become much more powerful, and since Windows ME/2000/XP latencies far lower can be used. SAWStudio and Sonar allowed to use such low settings from the start. Sequoia was updated in version 5.91, WaveLab in version 3.04.

In the Hammerfall's Settings dialog the MME buffersize (in fact the DMA buffer size) is set with the same buttons as the ASIO buffer size. Our test computers allow to use settings down to 64 samples without clicks. Please note that this setting only defines the buffer size 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 case can happen unnoticed when using ASIO and MME at the same time (multi-client) and setting ASIO to 186 ms, while the buffers in the MME application are still set for a lower latency. Playback will be stuttering and audio will be distorted.

Example: when you set the Hammerfall to 512 you can't use 128 in any program. But setting DMA to 128 allows 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 check out by yourself which lowest setting your system and software allows. Some motherboards with insufficient PCI bandwidth (especially VIA based) suffer from crackling at settings below 512. Be sure to set the buffer size to 512 or higher in such a case.

9. Configuring the Hammerfall

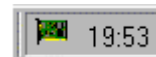
9.1 General Information

Configuring the Hammerfall is done using its own settings dialog. The panel 'Settings' can be opened in three different ways:

- by clicking on the hammer icon in the Taskbar's system tray
- by starting the 'Hammerfall' link from the Desktop
- via 'shortcut key' as defined in the 'Digi9652' link (default: Ctrl-Num2)



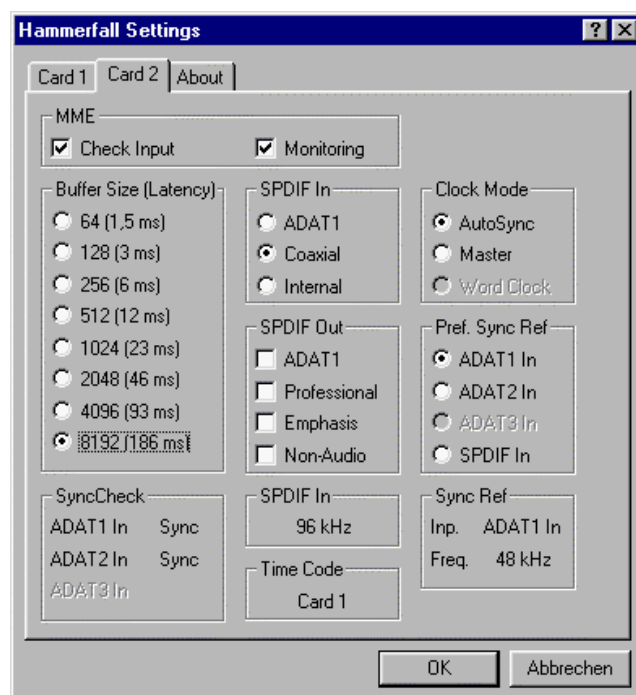
Under Windows 2000/XP Settings can be activated by clicking on the DIGI96 symbol in the Taskbar.



The DIGI9636 hardware offers a number of helpful, well thought-of practical functions and options which affect how the card operates - the DIGI9636 can be configured to suit many different requirements. The following is available in the 'Settings' dialogue:

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

Any changes made in the Settings dialog are applied immediately - confirmation (e.g. by clicking on OK or exiting the dialogue) is not required. However, settings should not be changed during playback or record if it can be avoided, as this can cause unwanted noises. 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. In general, we recommend disabling 'Audio active in background' (assuming this option is available).



The status displays at the bottom of the dialog box give the user precise information about the current status of the board, 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 'Sync Ref' display shows the input and frequency of the current sync source.

'Time Code' displays time information received from the card's ADAT Sync In. 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 record will always be allowed, even with non-valid input signals.

Monitoring activates the automatic pass-through of the input signal when in record mode. Both settings are valid for MME only.

AEB

After activating *ADAT1 Int.* the optical input ADAT1 will be routed to the internal input (CD In). Then the 4- or 8-channel signal of an AEB-I can be received. The optical input can no longer be used with an ADAT signal, but can still be used as SPDIF input. For this to work select 'ADAT1' under 'SPDIF In'.

Buffer Size

The setting *Buffer Size* determines the latency between incoming and outgoing ASIO* data, as well as affecting system stability (see chapter 13). Under Windows MME this setting determines the DMA buffer size (see chapter 8.6).

SPDIF In

Defines the input for the SPDIF signal. 'Coaxial' relates to the white phono plug, 'Internal' to the jumper CD In (ST3), 'ADAT1' to the optical input ADAT1.

SPDIF Out

The SPDIF output signal is constantly available at the internal jumper Sync Out (ST4) and the red 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 chapter 12.

Clock Mode

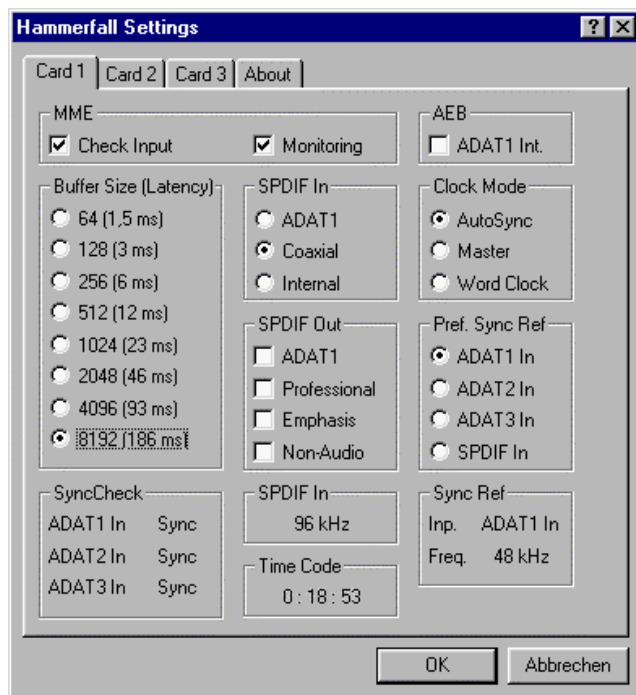
The card can be configured to use the following clock sources: external input signal (AutoSync), internal clock (Master), or external word clock signal (Wordclock).

Pref. Sync Ref.

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 is displayed in the *SyncRef* display.

Status Displays

The status displays at the bottom of the dialog box give various useful information, like synchronicity of the inputs, sample rate at the SPDIF input, detected Time Code at the ADAT Sync In, and input and frequency of the current sync source.



* Under W2k/XP also GSIF data


Note: Screenshot shows DIGI9652 settings dialog

9.2 Clock Modes - Synchronization

In the digital world, all devices are either the 'Master' (clock source) or a 'Slave' synchronized to the master. Whenever several devices are linked within a system, there must always be a single master clock. The Hammerfall's intelligent clock control is very user-friendly. Selecting 'AutoSync' will activate this mode.

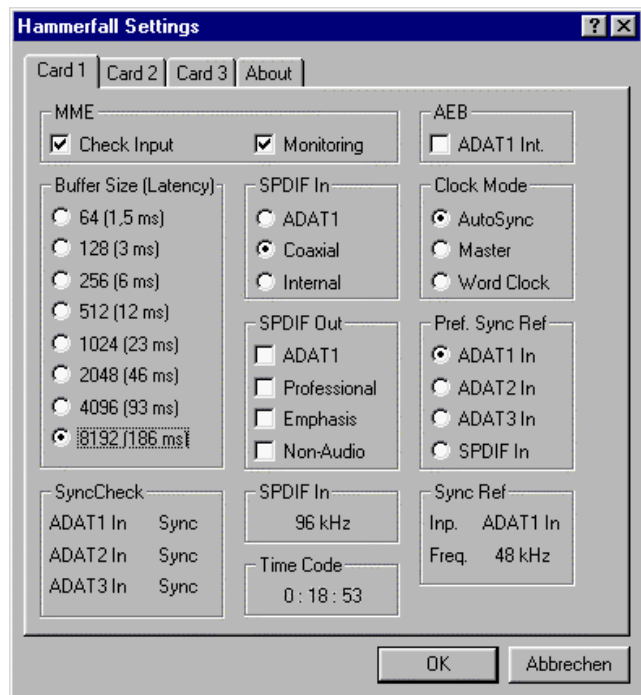
In AutoSync mode, the Hammerfall constantly scans all digital inputs for a valid signal. If this signal corresponds with the current playback sample rate, the card switches from the internal quartz (Sync Ref displays 'Internal') to a clock generated from the input signal (Sync Ref displays 'SPDIF' or '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 Hammerfall, AutoSync causes feedback in the digital carrier, so synchronization breaks down. To remedy this, switch the card's clock mode over to 'Master'.

 Remember that a digital system can only have one master! If the Hammerfall's clock mode is set to 'Master', all other devices must be set to 'Slave'.

All the ADAT optical inputs in the DIGI9636 as well as the SPDIF input will work simultaneously. Because there is no input selector however, the DIGI9636 has to be told which of the signals is the sync reference (a digital device can only be clocked from a *single* source). This is why the card has been equipped with automatic clock source selection, which adopts the first available input with a valid digital signal as the clock reference input. The input currently used as sync reference is shown in the 'Sync Ref' status field, together with the current sample frequency.

Via 'Pref Sync Ref' (preferred synchronization reference) a preferred input can be defined. As long as the card sees 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'.



To cope with some situations which may arise in studio practice, setting 'Pref Sync Ref' is essential. One example: An ADAT recorder is connected to the ADAT1 input (ADAT1 immediately becomes the sync source) and a CD player is connected to the S/PDIF 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) clock from the ADAT i.e. out of sync. In this case, 'Pref Sync Ref' should be temporarily set to SPDIF.

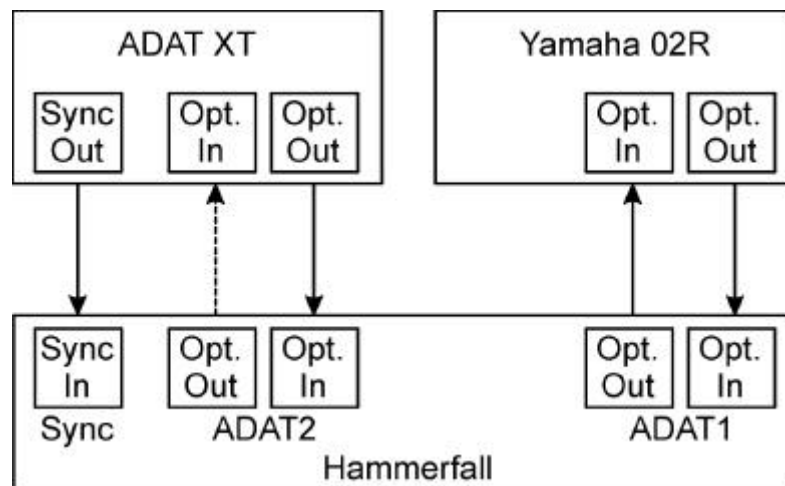
Note: Screenshot shows DIGI9652 settings dialog

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. RME's exclusive *SyncCheck* technology, first implemented in the Hammerfall, enables an easy to use check and display of 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 three ADAT optical inputs. The 'Sync Ref' display shows the current sync source's input and frequency.

In practice, SyncCheck provides the user with an easy way of checking whether all digital devices connected to the system are properly configured. With SyncCheck, finally anyone 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 input is receiving a signal from a digital mixing desk that has been set to clock mode 'Internal' or 'Master'. An ADAT recorder is connected to the ADAT2 input. The DIGI9636 is set to AutoSync mode. SyncCheck will now show that ADAT1 and ADAT2 inputs are no in sync, the display shows 'Lock' instead of 'Sync' for the ADAT2 input. Because the ADAT recorder is not receiving any signal from Hammerfall or

from the mixer, it will generate its own clock at a rate which is (almost) the same as the sample frequency of the mixing desk - but not identical. Remedy: To drive the ADAT recorder from its digital input, set it to slave mode (DIG), and connect the input to the Hammerfall's ADAT2 output. DIGI9636 is already in sync with the mixing desk, so it will send an identical (synchronous) signal to



ADAT2 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 technique and a lightning fast PLL, the DIGI9636 is not only capable of handling standard frequencies, but also any sample rate between 25 and 105 kHz. The input selected in 'Pref Sync Ref' serves as synchronization source. If the Expansion Board has been installed, and the word clock input is selected (clock mode 'Word Clock'), this will serve as the synchronization source, allowing any sample frequency between 25 kHz and 56 kHz in varispeed operation.

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

At 88.2 or 96 kHz: If one of the ADAT inputs has been selected in 'Pref Sync Ref', the sample frequency shown in the 'SPDIF In' field differs from the one shown in 'Sync Ref'. The card automatically switches to its Sample Split mode here, because ADAT optical inputs and outputs are only specified up to 48 kHz. Data from/to a single input/output is spread over two channels, the internal frequency stays at 44.1 or 48 kHz. In such cases, the ADAT sample frequency is only half the SPDIF frequency.

10. Word Clock

10.1 Technical Description and Usage

Correct interpretation of digital audio data is dependent upon a definite sample frequency. Signals can only be correctly processed or transferred between devices if these all share the same clock, otherwise digital signals are misinterpreted, causing distortion, clicks/crackle and even dropouts.

AES/EBU, SPDIF and ADAT are self-clocking, so an additional line for word clock could 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 without there being a defined 'master' device, self-clocking may break down. Besides, the clocks of all devices must be synchronized from a single source. Devices without SPDIF inputs (typically playback devices such as CD players) cannot be synchronized via self-clocking.

In digital studios, synchronization requirements can be met by connecting all devices to a central sync source. For instance, the master device could be a mixing desk, sending a reference signal - word clock - to all other devices. However, this will only work if all the other devices have word clock inputs (e.g. some professional CD players) allowing them to run as slaves. This being the case, all devices will receive the same clock signal, so there is no fundamental reason for sync problems when they are connected together.

10.2 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 the 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.


In practice, the situation has improved in recent years. The relatively low frequency of word clock signals is not a problem for modern electronic circuits. Because of the higher voltage, word clock networks are often more stable and reliable if cables are not terminated at all. Also, 75 Ohm cable is almost impossible to find these days. 50 Ohm cable is standard - this will also work as long as the termination resistors are 75 Ohm.

The word clock input on Hammerfall's Expansion Board is a high-impedance type ensuring maximum flexibility, and is therefore not terminated. If normal termination is necessary (e.g. because Hammerfall is the last device in the chain), simply connect a T-adapter to its BNC input jack, connect the cable supplying the word clock signal to one arm of the T-adapter and terminate the other with a 75 Ohm resistor (as a short BNC plug).

In case the Hammerfall resides within a chain of devices receiving word clock, plug a T-adapter into Hammerfall's BNC input jack and the cable supplying the word clock signal to one end of the adapter (as above), but connect the free end to the next device in the chain via a further BNC cable. The last device in the chain should be terminated using another T-adapter and a terminator plug as described in the previous paragraph.

10.3 General Operation

The green 'Lock' LED next to the input jack will light up when the input sees 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 there is a valid signal at the BNC jack, 'Sync Ref' will display 'Word'. This message has the same function as the green 'Lock' LED next to the BNC input jack, 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 wordclock input and 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.*

11. Using more than one Hammerfall

The current drivers support any combination and number of Hammerfalls, both DIGI9636 and DIGI9652. Please note that only one ADAT Sync can be used (of course). Additional all cards must be in sync i.e. have to receive valid sync information (either via wordclock or using AutoSync).

12. Special Characteristics of the SPDIF Output

Apart from the audio data itself, digital signals in SPDIF or AES/EBU format have a header containing channel status information. False channel status is a common cause of malfunction. The Hammerfall 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 always be played back with the emphasis bit set!*

This can be done by selecting the 'Emphasis' switch in the Settings dialogue ('SPDIF Out'). This setting is updated immediately, even during playback. The Hammerfall'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 Hammerfall 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 section 8.1 'Connections'), but with a male XLR plug instead of a female one.

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

The audio bit in the header can 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.

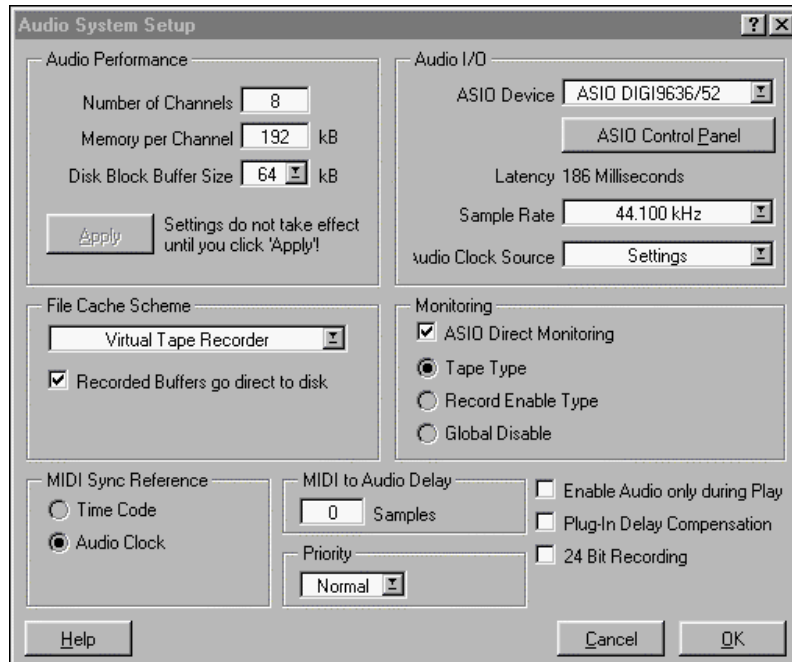
13. Operation under ASIO 2.0

13.1 General

We will use Steinberg's Cubase VST as an example throughout this chapter. All information provided can easily be adapted to other programs.

Start the ASIO software and select 'System' from the Audio menu. Select 'ASIO DIGI9636/52' as the audio I/O device. The 'ASIO system control' button opens the Hammerfall's Settings dialog (see chapter 9, Configuration).

Hammerfall also allows simultaneous record and playback of SPDIF audio data together with record and playback in ADAT format. Please note that the external SPDIF devices have to be running in sync, otherwise recordings will be corrupted.



Project Hammerfall supports 'ASIO Direct Monitoring'. Please note that in this mode neither routing nor pan are supported. Therefore the input signals will only be routed to the same output channel. Other VST mixer settings have no effect. Additionally, as the Hammerfall has no internal digital mixer, the playback signal of the ADM channel will be replaced by the record signal.

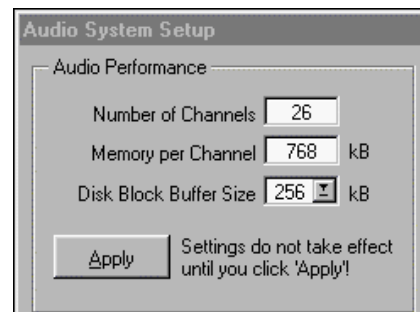
When the sample frequency is set to 88.2 or 96 kHz, all the ADAT optical inputs and outputs operate in Sample Split mode, so the number of available channels is reduced from 16 to 8.

13.2 Performance

The 'Audio Performance' settings are especially important. Firstly, the number of channels should be changed from 8 to 18 so that all the Hammerfall's inputs can be accessed.

A very common problem is insufficient hard disk performance. If the first track is missing while recording multiple tracks, or the error message 'Audio: Record Error' appears, the disk sub-system is too slow i.e. it is unable to write the audio data to the disk quickly enough. The problem can almost always be remedied 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. More tracks are only possible after changing 'Disk Block Buffer Size' to 256kB (depending on your computer). Please note that these parameters are only updated after clicking on 'Apply'.



The heyday of (expensive) SCSI hard disks in high-speed audio workstations is over. Today's cheap high-capacity EIDE disks allow continuous transfer rates of well over 10 MByte per second. In practical terms, this is more than enough to record up to 24 simultaneous tracks using Cubase and Hammerfall!

However, the hard disks have to work using 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.

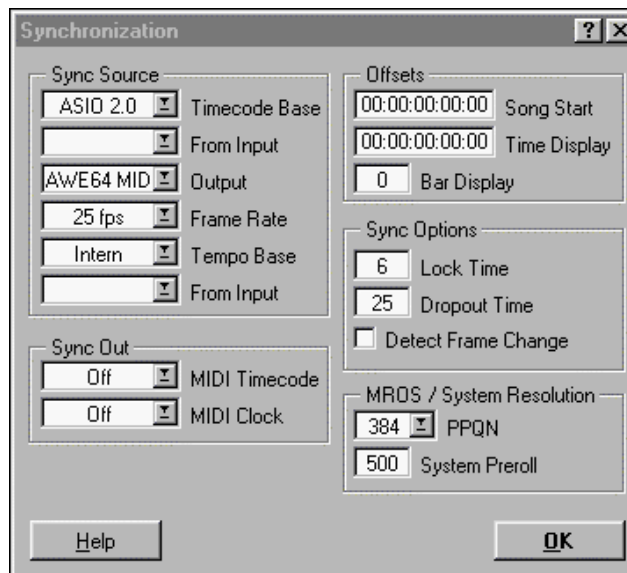
The Buffer Size value in Hammerfall's Settings dialog determines the latency (in this case the delay) between the audio application and the Hammerfall as well as general system stability. The higher the value, the more tracks can be recorded and played back simultaneously and the longer the system takes 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.

Present systems are mostly 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, 186 ms.

13.3 Synchronization

To achieve sample-accuracy between the ADAT recorder and Hammerfall while running Cubase, connect the ADAT sync output with the 9-pin D-type sync input of the DIGI9636. The 'Time Code' field in the Settings dialogue should now show the same position as the ADAT recorder.

Double-clicking on the Sync button in Cubase's transport panel will open the 'Synchronization' dialogue. Select ASIO 2.0 as the timecode base (under Sync Source), confirm the dialogue with 'OK', then activate Sync mode by (single) clicking on the Sync button.



If synchronization is not working i.e. Cubase does not respond when the ADAT is set to 'Play', please try the following:

- Check the cables
- Switch Sync off and on again (in Cubase's transport panel)
- Select 'Reset Devices' from the Options menu.
- Switch on the ADAT recorder(s) before starting Cubase
- Use the BRC as Master and send its word clock to all other devices

13.4 Known Problems

In case the used computer has no sufficient CPU-power and/or sufficient PCI-bus transfer rates, then drop outs, crackling and noise will appear. We also recommend to deactivate all Plugins to verify that these are not the reason for such effects.

Unfortunately some newer UltraATA66 and UltraATA100 hard disk controller (also Raid controller) seem to violate against the PCI specs. 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) are heard. Try to solve this problem by changing the default setting of the controller (for example by reducing the 'PCI Bus Utilization').

Another common source of trouble is incorrect synchronization. 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 Hammerfall must 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!

14. Operation under GSIF (Gigasampler Interface)

14.1 Windows 98/SE/ME

The GSIF interface of the Hammerfall's Windows 98/SE/ME driver allows direct operation with Gigasampler and Gigastudio, with up to 18 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, and so on.

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

Please note: The Hammerfall requires identical formats when used in multi-client operation! All programs simultaneously accessing the hardware MUST use the same sample rate.



Simultaneous operation of GSIF and ASIO requires to use different channels. As Cubase VST always uses tracks 1/2 these tracks must not be activated in Gigastudio/Sampler. Additionally the tracks activated in Gigastudio/Sampler have to be de-activated in the ASIO software (for example in the Master Bus section of Cubase).

14.2 Windows 2000/XP

Basically as under Windows 9x. Differences: GSIF under W2k/XP uses a modified interface, which needs interrupts (similar to ASIO). Therefore the user can now set and change the latency (under W9x latency was fixed inside Gigastudio). However, when using the Hammerfall, the latency is always the same as the one 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 W2k/XP driver fully supports multi-client operation, including the combination MME/ASIO. So for example Cubase, Gigastudio and Sonar can be used simultaneously, provided each of these programs uses its own audio channels exclusively. Please also note that Gigastudio is running unexpectedly 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.

15. Hotline - Troubleshooting

15.1 General

The newest information can always be found on our website www.rme-audio.com, section FAQ, Latest Additions.

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. Remedy: 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 cannot 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

- SPDIF 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 SPDIF output (channels 17 and 18).

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 Hammerfall 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.
- Activate Busmaster mode for the hard disks (see section 13.2 'Performance').
- 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 has to 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.

32 kHz files will not play back:

- Hammerfall does not have an internal 32 kHz clock, as this frequency is not covered by the ADAT standard. However, you can record and playback via SPDIF if the card is clocked from an external device i.e. AutoSync or Word Clock are active, and fed SPDIF or equivalent word clock signal is 32 kHz.

15.2 Installation

More information on installation problems (which fortunately are very seldom, thanks to Plug and Play), can be found in the Tech Info 'Installation problems'. It is located in the directory **\rmeaudio.web\techinfo** on the RME Driver CD.

Hammerfall is normally found in the Device Manager (>Settings/Control Panel/System<), category 'Sound-, Video- and Gamecontroller'. A double click on 'DIGI9636' starts the properties dialog. Choosing 'Resources' shows Interrupt and Memory Range.

The newest information on hardware problems can always be found on our website www.rme-audio.com, section FAQ, Hardware Alert: about incompatible hardware.

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

- Remove the optical cables from the DIGI9636 and check whether the optical outputs light up when the PC is switched on. If not, the card is either defective or not correctly inserted in the PCI slot.

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

- Check whether the Hammerfall appears in the Device Manager. If the 'DIGI9636' 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 Hammerfall has been selected as current ASIO device.

The computer crashes whenever the Hammerfall 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 Hammerfall via /Resources/Change Setting. Detailed information on this subject can be found in **\rmeaudio.web\techinfo\install.htm** on the RME Driver CD.

16. DIGICheck : Analysis, Tests and Measurements using the DIGI9636

The DIGICheck software is a unique utility developed for testing, measuring and analysing digital audio streams. DIGICheck is a subset of functions included in our well-known DAM-1 (Digital Audio Monitor). DIGICheck is not capable of all the DAM-1 functions, because the latter uses a built-in DSP for its calculations. By its very nature, DIGICheck uses up processor time, while the DAM-1 unit will run perfectly in the background.

Although the DIGICheck software is fairly self-explanatory, it still includes a comprehensive online help. A detailed description of all functions is also available in HTML format (digich.htm, in the **techinfo** directory on the RME Driver CD or from our website). The following is a short summary of the available functions:

- **Level Meter.** High precision 24-bit resolution, 2/8/18 channels. Application examples: Peak level measurement, RMS level measurement, over-detection, phase correlation measurement, dynamic range and signal-to-noise ratios, RMS to peak difference (loudness), long term peak measurement, input check.
- **Bit Statistics.** Shows the true resolution of audio signals as well as errors and DC offset.
- **Performance Test.** Measures overall system performance (computer + DIGI9636)
- **Memory Test.** Tests the DIGI hardware and the entire audio data path in the PC

DIGICheck is suitable for any card in the DIGI96 series. However, the Hammerfall does not support Channel Status data, which is why the *Channel Status Display* is not available.

To install DIGICheck, go to the **\DIGICheck** directory on the RME Driver CD and run *setup.exe*. Follow the instructions prompted on the screen.

17. Accessories

RME offers several optional components, further increasing the flexibility and usability of the Hammerfall system. Additionally parts of the Hammerfall, like the special adapter cable, are available separately.

Part Number	Description
36003	Optical cable, Toslink, 0.5 m (1.5 ft)
36004	Optical cable, Toslink, 1 m (3.3 ft)
36006	Optical cable, Toslink, 2 m (6.6 ft)
36007	Optical cable, Toslink, 3 m (9.9 ft)
36008	Optical cable, Toslink, 5 m (16.4 ft)
36009	Optical cable, Toslink, 10 m (32.8 ft)

Standard lightpipe with TOSLINK connectors, RME approved quality.

36018	Adapter cable D-sub to D-sub / 2 x phono
33045	Internal flat cable to Expansion Board
31143	Eprom W52-G Board Rev 1.1
31144	Eprom W36-G Board Rev 1.1
31145	Eprom M52-2 Board Rev 1.5/1.6
31146	Eprom M36-2 Board Rev 1.5/1.6
31161	Eprom W52-2 Board Rev 1.5/1.6
31204	Eprom W36-2 Board Rev 1.5/1.6

18. TECH INFO

See <http://www.rme-audio.com/techinfo/index.htm> or the directory **lrmeaudio.webltechinfo** on the RME Driver CD for more detailed information. At the time of writing, the following *Tech Info* is available:

Synchronization II (DIGI96 series)

Digital audio synchronization - technical background and pitfalls.

Installation problems

Problem descriptions and solutions.

Information on driver updates

Lists all changes in the drivers of the Hammerfall series.

Configuring Logic, Samplitude, Cubase, Cakewalk, Sonar and SAWPlus32

Step by step instructions for use with RME cards

DIGICheck: Analysis, tests and measurements with the DIGI96 series

A description of DIGICheck, including technical basics.

ADI-8 Inside

Technical information about the RME ADI-8 (24-bit AD/DA converter).

19. Warranty

Each individual Hammerfall undergoes comprehensive quality control and a complete test in a PC environment at RME before shipping. This may cause very slight signs of wear on the contacts (if the card looks like it was used one time before - it was). The usage of high grade components allows us to offer a full two year warranty. We accept a copy of the sales receipt as valid warranty legitimation.

RME'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 - replacement or repair in such cases can only be carried out at the owner's expense.

RME does not accept claims for damages of any kind, especially consequential damage. Liability is limited to the value of the Hammerfall. The general terms of business drawn up by Synthax OHG apply at all times.

20. Appendix

RME news, driver updates and further product information are available on our website:

<http://www.rme-audio.com>

If you prefer to read the information off-line, you can load a complete copy of the RME website from the RME Driver CD (in the **\rmeaudio.web** directory) into your browser.

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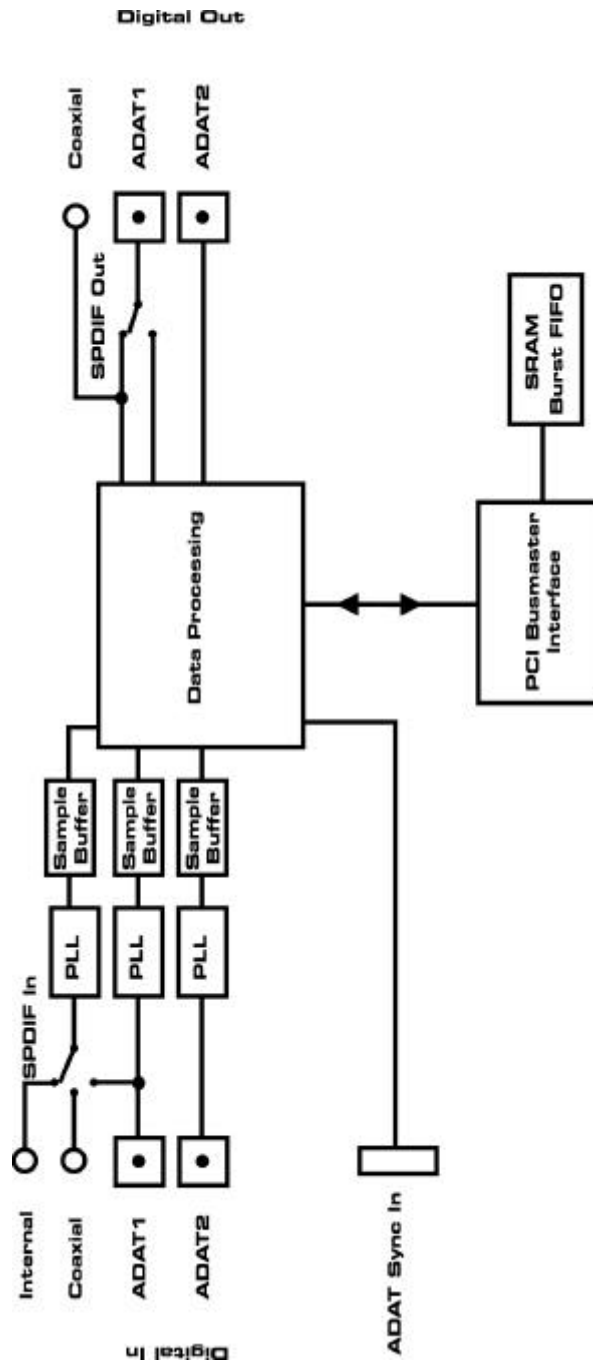
Copyright © Matthias Carstens, 6/2002. Version 2.0
Current driver version: W98: 2.60, NT: 3.86, W2k: 2.02

This manual applies to board rev. 1.5, hardware version 008.

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21. Diagrams

21.1 Block Diagram



21.2 Pin assignment of the cable adapter

Pin	Signal	Pin	Signal
9	SPDIF In +	5	SPDIF In -
6	SPDIF Out +	1	SPDIF Out -

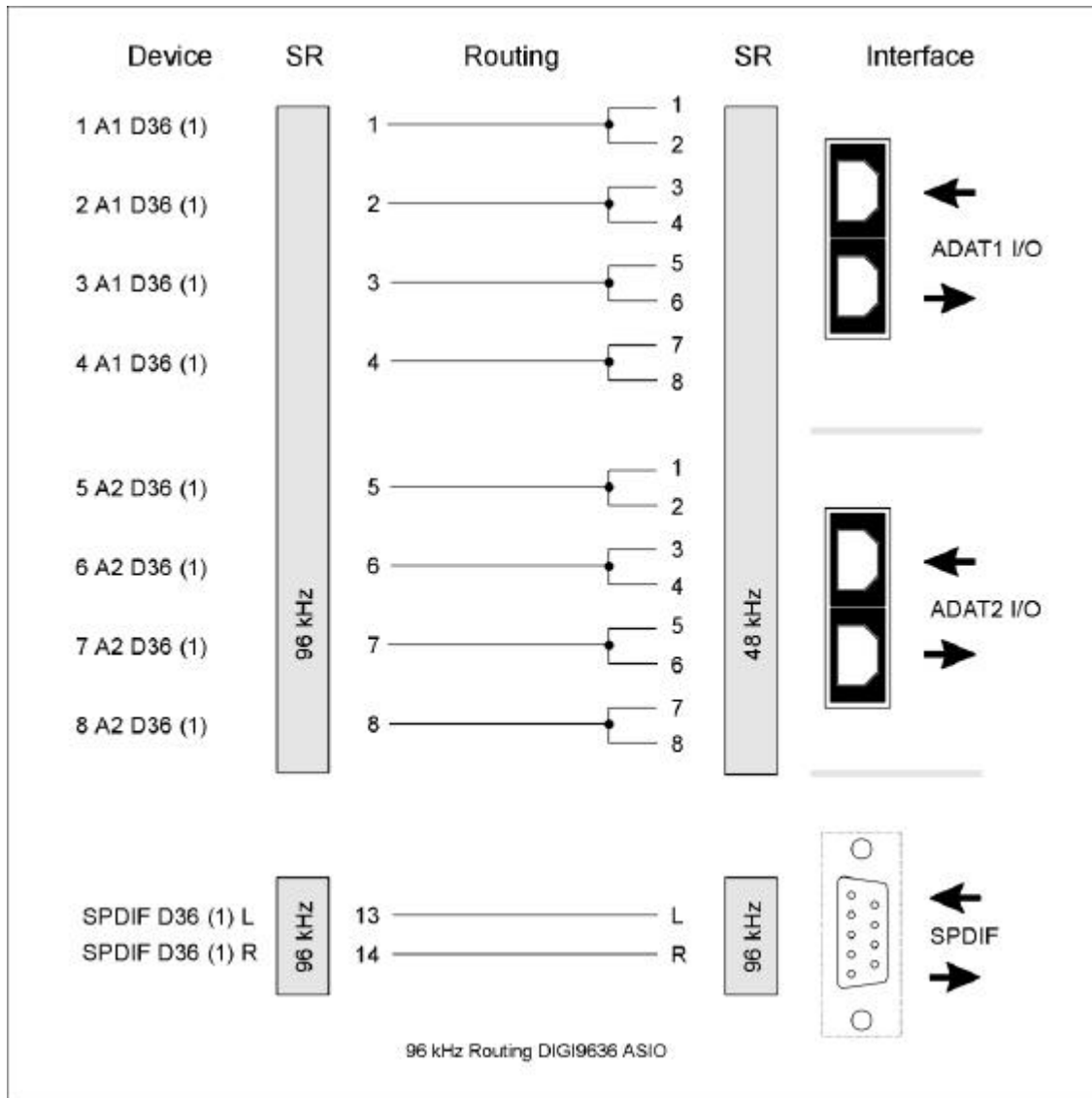
21.3 ADAT Track Routing, ASIO at 96 kHz

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.

Device: The device name in the audio application

SR: Sample Rate

Device name code: Channel in ASIO host, ADAT interface, DIGI9636, card number

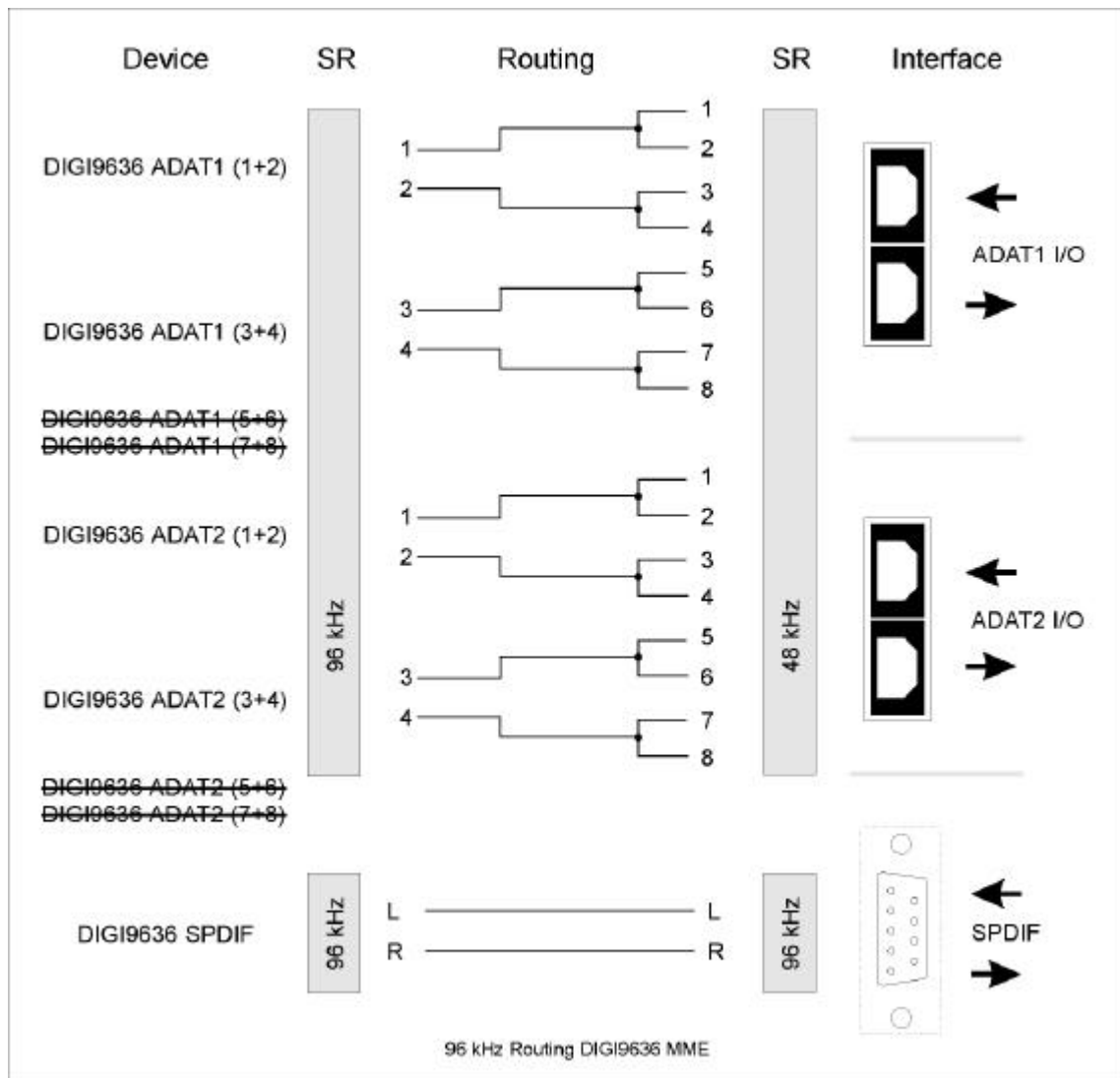


21.4 ADAT Track Routing, MME at 96 kHz

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



22. 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.