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The **ASIO Hardware**
0% CPU load at full
52 track operation

User's Guide - Macintosh Version



DIGI9636



ready



SyncAlign[®]

ZLM[®]



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

Board Rev. 1.5, Hardware Version 008

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1. Introduction

Thank you for choosing the RME 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 S/PDIF, AES/EBU or ADAT optical. The numerous unique features, well thought-out configuration dialog and unsurpassed low latency operation puts the DIGI9636 at the very top of the range of digital audio interface cards.

The package includes ASIO 2.0 drivers for MacOS. An ALSA driver for Linux or Unix is also available.

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

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)
- Lightpipe cable (2 x 2m)

3. System Requirements

- MacOS 8.x or higher. G3 or higher recommended.
- A free PCI rev. 2.1 Busmaster slot

4. Brief Description and Characteristics

- PCI Busmaster interface with additional burst FIFO
- No risk of dropouts, even at high PCI bus load
- ASIO design: 0% (zero!) CPU load, even using all 36 channels
- 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 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

5. Technical Specifications

5.1 Digital

- Low jitter S/PDIF: < 5 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 (word clock) 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
- Formats: S/PDIF (Consumer and Professional), ADAT optical

5.3 Transfer Modes: Resolution / Bits per Sample

- 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.

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

First fit the card (see 6. Hardware Installation), then switch on the computer and install the drivers from the RME Driver CD. The driver files are located in the folder 'Hammerfall'.

In case a newer driver version was downloaded from the RME website double-click the 'm9652_x.sit' archive to decompress it into separate files (using 'Aladin Stuffit Expander').



If you already installed an older version of the driver first make sure to remove all old files. To do so open the 'Extensions' folder which is inside your 'System' folder. Remove the file 'HAMMERFALL DRIVER'. Also remove 'HAMMERFALL SETTINGS' and/or 'RME DIGI Settings' from the directory where it was copied to. Remove the 'HAMMERFALL ASIO' driver file from any 'ASIO Drivers' folder.

Drag the new file **Hammerfall Driver** to the System folder. It will be installed automatically into the 'Extension' folder. Confirm the system's message to complete the installation. Now the driver file should be found in the 'Extension' folder, see example to the right.



Copy the new files **RME DIGI Settings** and **Hammerfall ASIO** into all 'ASIO Drivers' folders found on your computer. As every ASIO software has its own ASIO Drivers folder the files have to be copied several times. The picture to the left shows an example after installation of Hammerfall and DIGI96.

Configuration of the Hammerfall is done through the Settings dialog, which can be called from within any ASIO compatible software (for example Audio/System/ASIO Control Panel). To be able to call up the Settings dialog at any time we recommend to create an Alias on the desktop. To create an Alias select 'RME DIGI Settings' with the mouse cursor, press and hold the Apple and Alt keys on your keyboard, and drag 'RME DIGI Settings' to the desired location. With this all RME cards in the system can be configured easily even without starting an ASIO software.

To finish installation reboot the computer.

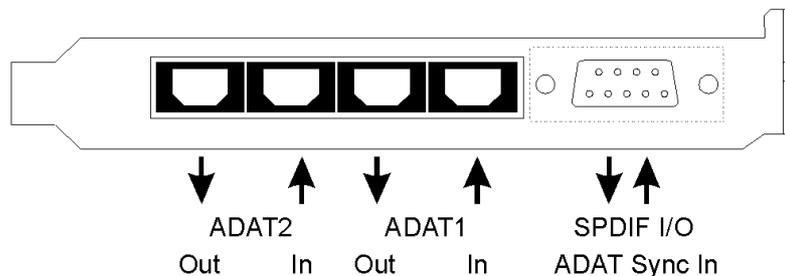
Linux/Unix

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 Connections

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/PDIF, if this mode is selected in the Settings dialog.



An input is selected via the RME DIGI Settings dialog. Hammerfall Light accepts the commonly used digital audio formats, S/PDIF as well as AES/EBU. Channel status and copy protection are ignored.

In S/PDIF 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, a cable adapter 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.

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 (advantage: direct transfer of data within the computer). It also allows to use an AEBx-I, when a 3-wire cable connects ST7 on AEB and Hammerfall.

Additionally the card offers two internal outputs, labeled ADAT1OUT (ST5) und 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.

9. Configuring the DIGI9636

9.1 General Information

Configuring Hammerfall Light is done using its own settings dialog, RME DIGI Settings.

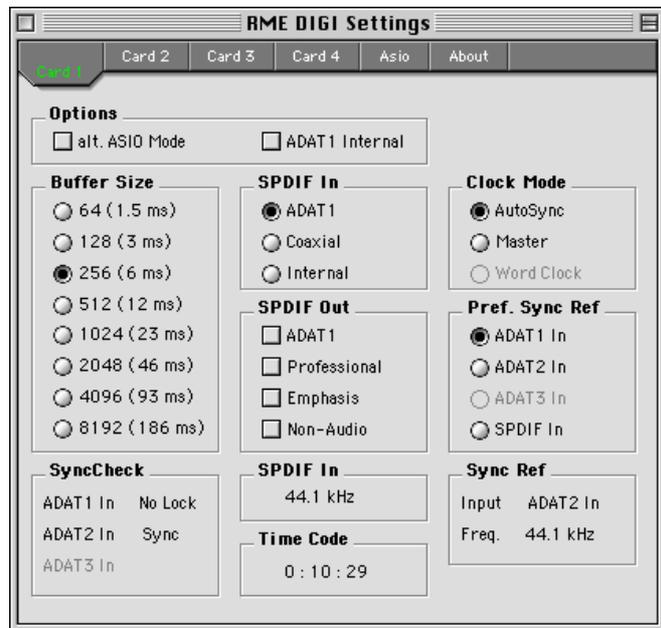
The DIGI9636 hardware offers a number of helpful, well thought-of practical functions and options which affect how the card operates - the DIGI9636 can be configured to suit many different requirements. The following is available in the 'Settings' dialog:

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

The setting 'Buffer Size' determines the latency between incoming and outgoing data, as well as affecting system stability (see chapter 13). We recommend selecting the highest value here (8192 Samples) - the board itself will still run comfortably.

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).



Buffer Size:

The setting 'Buffer Size' determines the latency between incoming and outgoing data, as well as affecting system stability (see chapter 13). We recommend selecting the highest value here (8192 samples) - the board itself will still run comfortably.

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 11.

Clock Mode:

The card can be configured to use the external input signal (AutoSync) or its internal clock (Master) as clock source.

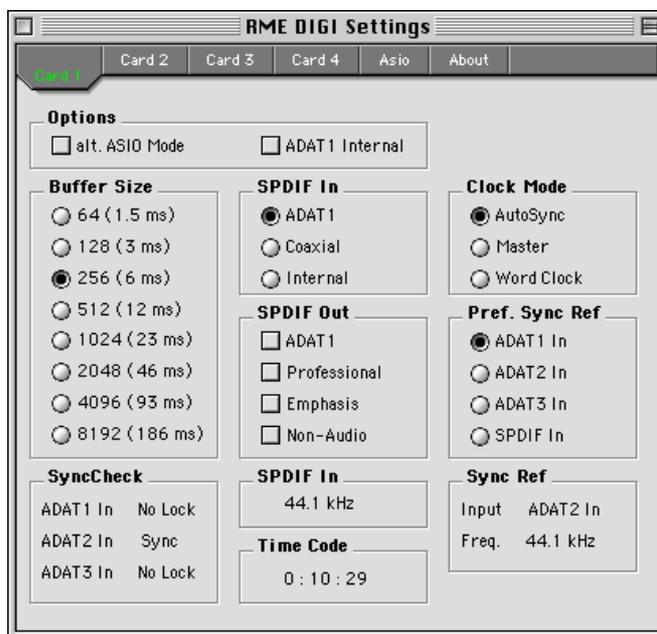
Pref. Sync Ref.:

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.

Options:

'Alt. ASIO Mode' activates a different ASIO callback method. This setting is performed in real-time and under operation. Therefore it's very easy to check whether this setting results in any performance advantages. This setting is recommended for Logic (emagic) and Spark (TC).

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'.



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 DIGI9636'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 DIGI9636 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 'S/PDIF' 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.

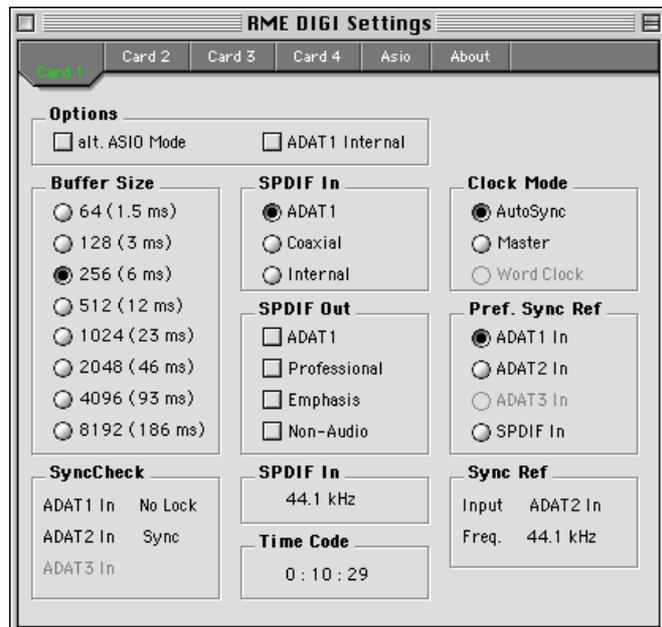
AutoSync guarantees that normal record and record-while-play will always work correctly. In certain cases however, e.g. when the inputs and outputs of a DAT machine are connected directly to the DIGI9636, AutoSync causes feedback in the digital carrier, so synchronization breaks down. To remedy this, switch the DIGI9636 clock mode over to 'Master'.

 Remember that a digital system can only have one master! If the DIGI9636'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 S/PDIF 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 S/PDIF.

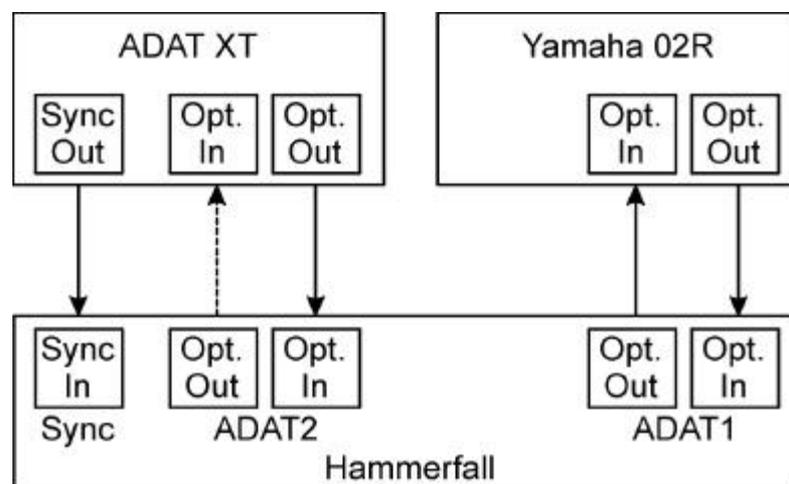


If several digital devices are to be used simultaneously in a system, they not only have to operate with the same sample frequency but also be synchronous with each other. This is why digital systems always need a single device defined as 'master', which sends the same clock signal to all the other ('slave') devices. RME's exclusive *SyncCheck* technology (first implemented in the DIGI9636) 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 and ADAT2 inputs are receiving 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 DIGI9636 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. Because the ADAT recorder is not receiving any signals 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 ADAT3 output. Hammerfall 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 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 'S/PDIF 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 'S/PDIF 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 S/PDIF frequency.

10. Using more than one Hammerfall

The current drivers support any combination and number of Hammerfalls, both DIGI9636 and DIGI9636. 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 word clock or using AutoSync).

11. Special Characteristics of the S/PDIF Output

Apart from the audio data itself, digital signals in S/PDIF or AES/EBU format have a header containing channel status information. False channel status is a common cause of malfunction. The DIGI9636 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 dialog ('S/PDIF Out'). This setting is updated immediately, even during playback. The DIGI9636'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 μ s
- Aux bits Audio Use

Professional AES/EBU equipment can be connected to the DIGI9636 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.

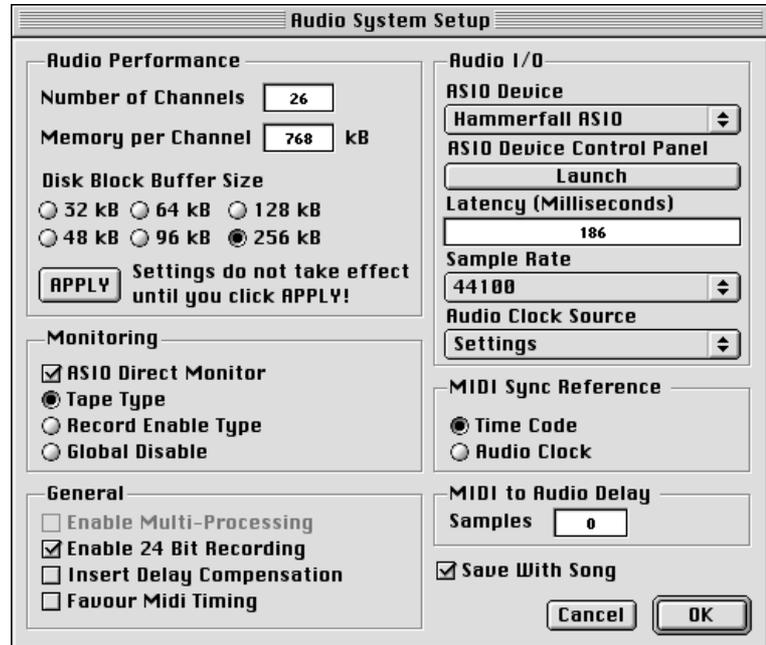
12. Operation under ASIO 2.0

12.1 General

At the time of writing, Cubase VST is the only available ASIO 2.0 application, so it will be used as an example throughout this chapter.

Start the ASIO software and select 'System' from the Audio menu. Select 'Hammerfall ASIO' as 'ASIO Device'. The button 'ASIO Device Control Panel' opens the DIGI9636 Settings dialog (see chapter 9, Configuration).

DIGI9636 also allows simultaneous record and playback of S/PDIF audio data together with record and playback in ADAT format. Please note that the external S/PDIF devices have to be running in sync, otherwise recordings will be corrupted.



Enhanced ZLM (Zero Latency Monitoring) in the DIGI9636 is so efficient that even different sample rates would usually cause no noticeable clicks or other unwanted noises. Having said this, you should not be tempted to 'misuse' this feature, because Murphy's Law is sure to ruin your 'take of the century'.

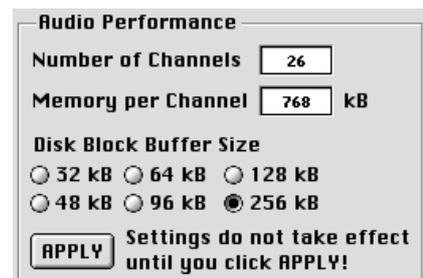
DIGI9636 supports 'ASIO Direct Monitoring'. Please note that in this mode neither routing nor pan are supported so the input signals will only be routed to the same output channel. Other VST mixer settings have no effect.

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. Some programs have problems to reset the ASIO driver properly. To change the number of ASIO channels it can be necessary to restart the program for the change to take effect.

12.2 Performance

The 'Audio Performance' settings are especially important. Firstly, the number of channels should be changed from 8 to 26 so that all the DIGI9636 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. 26 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'.

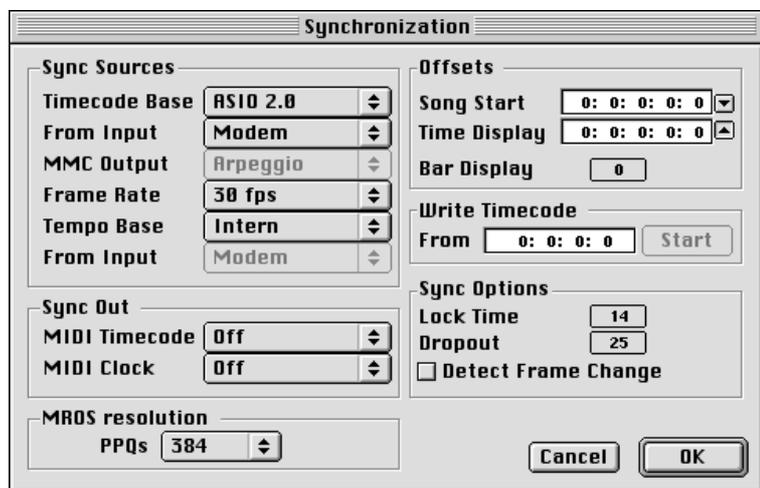
The heyday of (expensive) SCSI hard disks in high-speed audio workstations is over. Today's cheap high-capacity EIDE disks allow continuous transfer rates of well over 5 MB per second. In practical terms, this is more than enough to record up to 24 simultaneous tracks using Cubase and Hammerfall!

The Buffer Size value in Hammerfall's Settings dialog determines the latency (in this case the delay) between the audio application and the DIGI9636 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.

12.3 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 dialog should now show the time information as the ADAT recorder.

Double-clicking on the Sync button in Cubase' transport panel will open the 'Synchronization' dialog. Select ASIO 2.0 as the timecode base (under Sync Source), confirm the dialog 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' 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

12.4 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.

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 DIGI9636 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!

13. Compatibility

13.1 Software

The Hammerfall series is compatible to all major ASIO applications, like Cubase VST, emagic Logic, Opcode Studio Vision PRO, Prosoniq SonicWORX, TC SPARK, Peak from Bias, Motu Digital Performer and Super Collider.

13.2 Hardware

No problems known.

13.3 General PCI related problems

The following symptoms are typical for PCI related problems:

- The control panel of the Sound Manager is crossed red
- When booting the control panels are displayed too big, or spread 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).

14. Hotline

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 globally disabled.

The first 8 channels don't seem to work

- S/PDIF 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/PDIF output (channels 17 and 18).

The performance with emagic's Logic is poor

- Ensure that 'Alt.ASIO Mode' is checked (RME DIGI Settings, Options). This setting can be changed on the fly.

15. Troubleshooting

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 Mac 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 DIGI9636 has been selected as current ASIO device.

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 DIGI9636 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 card'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 card's clock mode to 'Master'.
- Increase the buffer size of the hard disk cache.

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 S/PDIF if the card is clocked from an external device i.e. AutoSync or Word Clock are active, and fed S/PDIF or equivalent word clock signal is 32 kHz.

16. TECH INFO

See <http://www.rme-audio.com/techinfo/index.htm> or the directory **rmeaudio.web\techinfo** 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.

Information on driver updates

Lists all changes in the drivers of the Hammerfall series.

ADI-1 Inside

Technical information about the RME ADI-1 (20-bit AD/DA converter).

ADI-8 Inside

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

TMS (Track Marker Support)

Description of the TMS technology to transfer CD- and DAT information.

17. Warranty

Each individual DIGI9636 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 DIGI9636. The general terms of business drawn up by Synthax OHG apply at all times.

18. 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 **lrmeaudio.web** directory) into your browser.

Distributor in Germany:
Synthax, Am Pfanderling 62, D-85778 Haimhausen, Tel.: (49) 08133 / 91810

Manufacturer:
Ingenieurbuero Mueller, Goethestr. 22, D-09648 Mittweida

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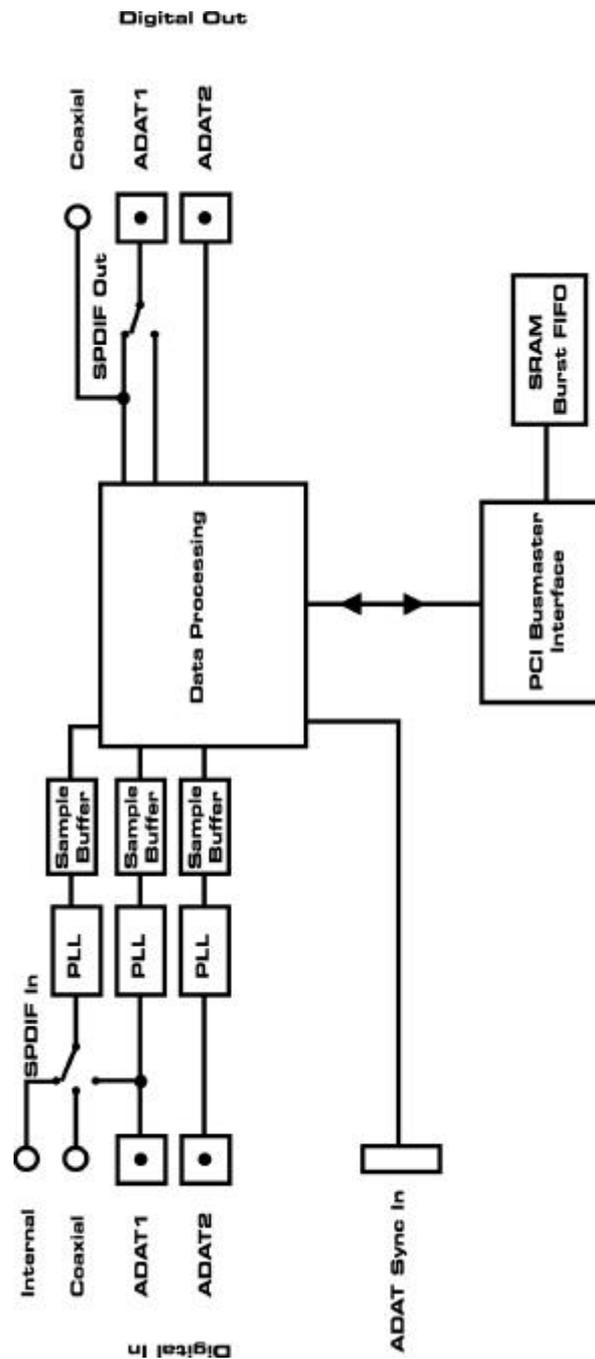
Copyright © Matthias Carstens, 08/00. Version 1.1
Current driver version: 1.4

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

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19. Diagrams

19.1 Block Diagram



19.2 Pin assignment of the cable adapter

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

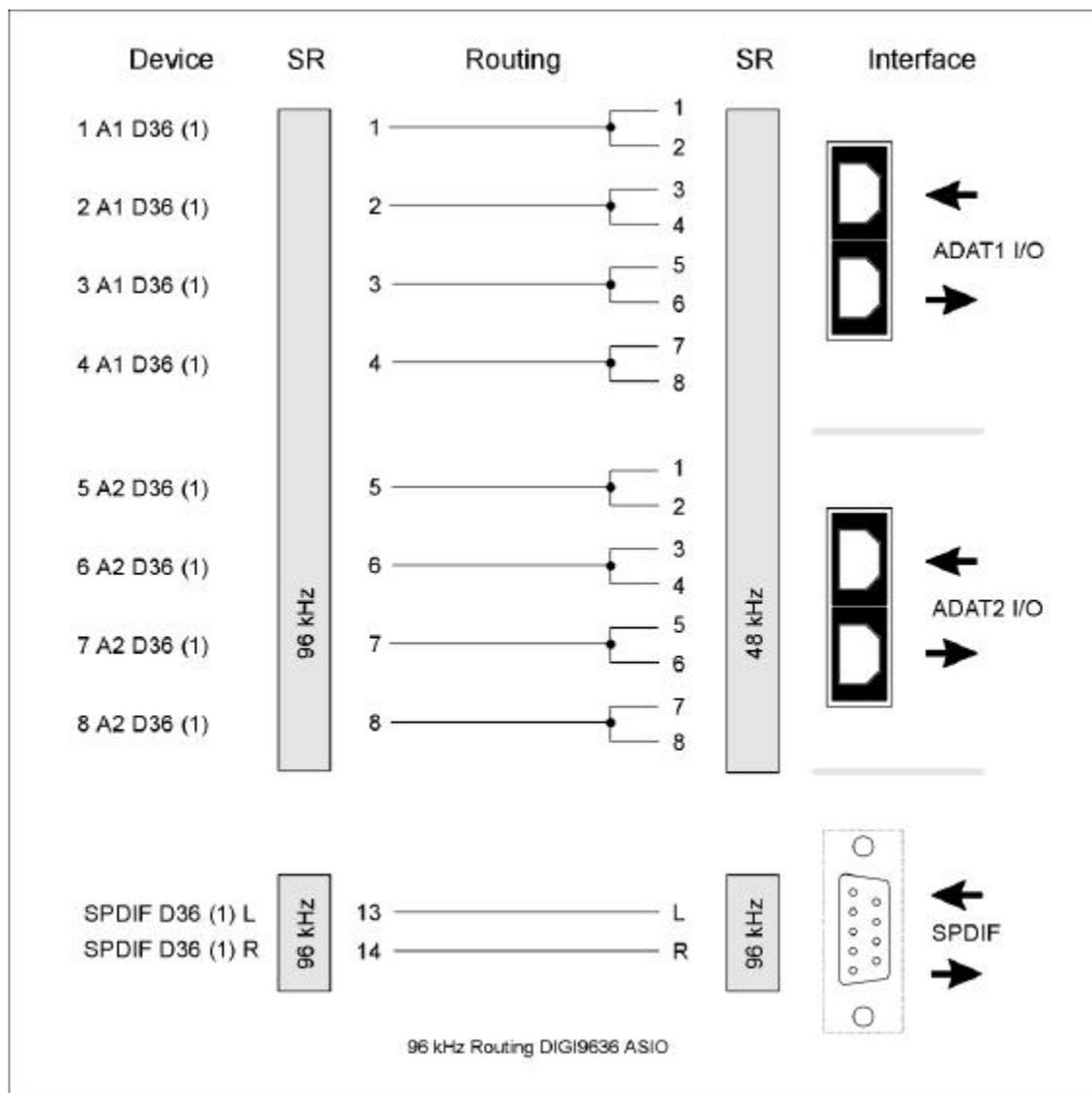
19.3 ADAT Track Routing, ASIO at 96 kHz

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



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.