User's Guide - Macintosh

Hammerfall® DSP System

HDSP 9652

TotalMix™

SyncAlign® ZLM® SyncCheck®

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

Contents

1. Introduction

Thank you for choosing the Hammerfall DSP. This unique audio system is capable of transferring digital audio data directly to a computer from practically any device equipped with a digital audio interface, be it SPDIF, AES/EBU or ADAT optical. The numerous unique features and well thought-out configuration dialog puts the Hammerfall DSP at the very top of the range of digital audio interface cards.

The package includes drivers for Windows 98/2000/XP and MacOS. An ALSA driver for Linux is planned to be available soon (see chapter 7).

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 Hammerfall DSP System's package contains each of the following:

- HDSP 9652 PCI card
- HDSP 9652 expansion board
- Quick Info guide
- RME Driver CD
- Adapter cable (D-type to D-type/phono)
- MIDI breakout cable
- Internal cable (2-core)
- Flat ribbon cable (14-core)
- 2 optical cable (TOSLINK), 2 m (6.6 ft)

3. System Requirements

- MacOS 8.6 or greater. G3 300 MHz recommended
- PCI Interface: a free PCI rev. 2.1 Busmaster slot
- CardBus Interface: a free PCMCIA Slot type II, CardBus-compatible

4. Brief Description and Characteristics

- Hammerfall design: 0% (zero!) CPU load, even using all 52 ASIO channels
- All settings can be changed in real-time
- Enhanced mixed mode: ADATs 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 12 channel, 96 kHz/24-bit record/playback via ADAT optical
- Slave and master clock modes
- Automatic and intelligent master/slave clock control
- Unsurpassed Bitclock PLL (audio synchronization) in ADAT mode
- Word clock input and output
- ADAT Sync in (9-pin D-type) for sample-accurate transfer
- Zero Latency Monitoring: Hardware bypass per track, controlled by Punch in/out
- Enhanced ZLM for latency-free submixes and perfect ASIO Direct Monitoring
- SyncAlign guarantees sample aligned and never swapping channels
- SyncCheck tests and reports the synchronization status of input signals
- 2 x MIDI I/O, 32 channels high-speed MIDI
- DIGICheck DSP: Level meter in hardware, peak- and RMS calculation
- TotalMix: 1352 channel mixer with 40 bit internal resolution

5. Technical Specifications

5.1 Digital

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

5.2 Digital Interface

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

5.3 MIDI

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

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

1. Disconnect the power cord and all other cables from the computer.
2. 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 HDSP 9652 from its protective bag, discharge any static in your body by touching the metal chassis of the PC.
4. Connect the HDSP 9652 card with the Expansion Board using the supplied flat ribbon cable. Note: The connectors on the cable cannot be plugged in the wrong way round.
5. Insert the HDSP 9652 firmly into a free PCI slot, press and fasten the screw.
6. Insert the Expansion Board and fasten the screw.
7. Replace the computer's housing.
8. Reconnect all cables including the power cord.

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

7. Software 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 on the CD in the folder 'Hammerfall DSP'.

In case a newer driver version was downloaded from RME's website double-click the 'madsp_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 DSP Driver'. Also remove 'Hammerfall DSP Settings' from the directory where it was copied to. Remove the 'Hammerfall DSP ASIO' driver file from any 'ASIO Drivers' folder.

After unstuffing the archive the driver files are found in folders. The name of the folders tell where to copy the files! The driver installation is done manually in **5** steps:

1. Drag the file **Hammerfall DSP Driver** from 'into System folder' into 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.

2. Copy the files **Hammerfall DSP Settings**, **Hammerfall DSP ASIO** and **Hammerfall DSP ASIO 96 kHz** from 'into ASIO Drivers folder' into all 'ASIO Drivers' folders found on your computer. As every ASIO software has its own ASIO Drivers folder, the files most propably have to be copied several times.

Configuration of the Hammerfall DSP 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 'Hammerfall DSP Settings' with the mouse cursor, press and hold the Apple and Alt keys on your keyboard, and drag **Hammerfall DSP Settings** to the desired location.

3. Copy the complete sub-folder **Hammerfall DSP**, found in the folder 'into Preferences folder', to the system folder 'Preferences'. This way the files related to the HDSP system reside in their own folder, without cluttering the Preferences folder. Additionally deleting those files is much easier in case of a driver update. **Hammerfall DSP** contains 10 files:

default.mix: Default settings for TotalMix
default.vol: Default settings for Digiface/Multiface, as long as TotalMix isn't started
preset1.mix to **preset8.mix**: Presets for the HDSP mixer

4. The file **Hammerfall DSP TotalMix** can be copied to any place. When started the HDSP mixer comes up and allows you to configure the digital real-time mixer of the Digiface.

TotalMix requires Carbon Library 1.1, which is part of the operating system since MacOS 9.1. After installation of Carbon Library 1.1 TotalMix can even be run on older systems (down to 8.6).

5. Using the MIDI ports of the Digiface requires an installed OMS (Open Music System) from Opcode. The latest version 2.3.8 can be downloaded for free at

http://www.opcode.com

After the installation of OMS, copy the file **HDSP_OMSDriver**, found in the folder 'into OMS Folder folder', into the system folder 'OMS Folder'.

To finish installation reboot the computer.

After re-boot the MIDI driver is installed, but not yet activated. To activate it create a new OMS Studio setup. Using 'Search' the MIDI driver of the Digiface should be found and added to the list. Now it can be activated.

Linux/Unix

An ALSA driver for Linux/Unix is planned to be available soon. Further information on ALSA is available at

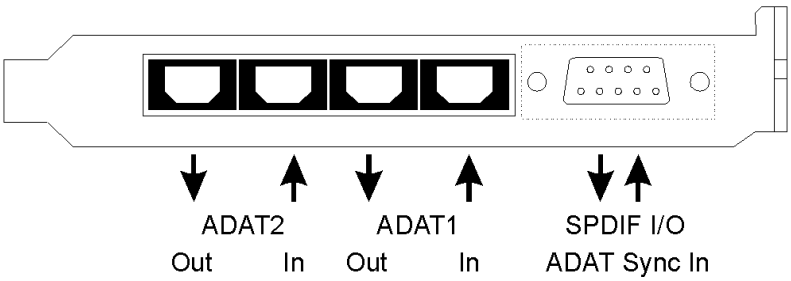
http://www.alsa-project.org

8. Operation and Usage

8.1 External Connections

HDSP 9652 consists of the main PCI board and an Expansion Board. All the essential electronics are located on the PCI card, so it will also work without the Expansion Board.

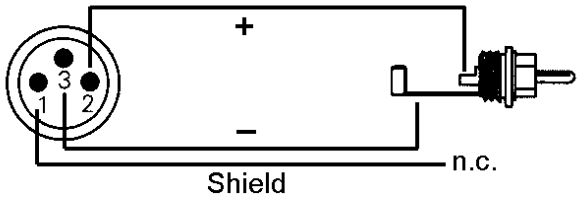
The main board's bracket has two ADAT optical inputs and two ADAT optical outputs, as well as a 9-pin D-type socket. Coaxial S/PDIF input and output requires plugging in the adapter cable, whereby the red phono socket is the output. The ADAT1 I/O next to the D-type socket can also be used for optical 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.

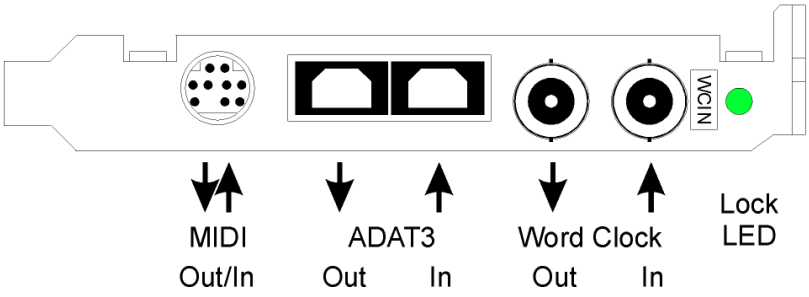
In SPDIF mode, identical signals are available at both the optical and the coaxial outputs. An obvious use for this would be simply connecting two devices, i.e. using the HDSP 9652 as a splitter.

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.

The Expansion Board's bracket gives access to a third ADAT optical input and output as well as word clock I/O. Next to the two BNC sockets is an LED, which displays the word clock input lock status. The included breakout cable is connected to the 9-pin Mini-DIN connector and realizes two MIDI inputs and outputs.



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.

This connector can also be used to connect an AEBx-I. Be sure to use a 3-wire cable to connect ST7 on both AEBx-I and Hammerfall too. Using jumper ST5 a second AEBx-I can be connected. Remove the jumper (position ADAT2 IN), and connect the 2-core cable of the AEBx-I to position 'AEB2 IN'. Please note the label GND for correct polarity. The optical input ADAT2 can't be used in this configuration.

Additionally HDSP 9652 offfers two internal outputs, labeled ADAT1OUT (ST4) and ADAT2OUT (ST1). 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.

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

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

8.3 Recording Digital

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 Hammerfall DSP system: a comprehensive I/O signal status display (showing sample frequency, lock and sync status) in the Settings dialog, and status LEDs for each input.

The sample frequency 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, Hammerfall DSP displays the current sample frequency. This parameter can then be changed in the application's audio attributes (or similar) dialogue.

It often makes sense to monitor the input signal or send it directly to the output. This can be done at zero latency using **TotalMix** (see chapter 14).

For an automated real-time monitoring function the HDSP System supports ASIO Direct Monitoring (ADM) in ASIO 2.0. When 'ASIO Direct Monitoring' has been switched on the input signal is routed in real-time to the output whenever Record is started.

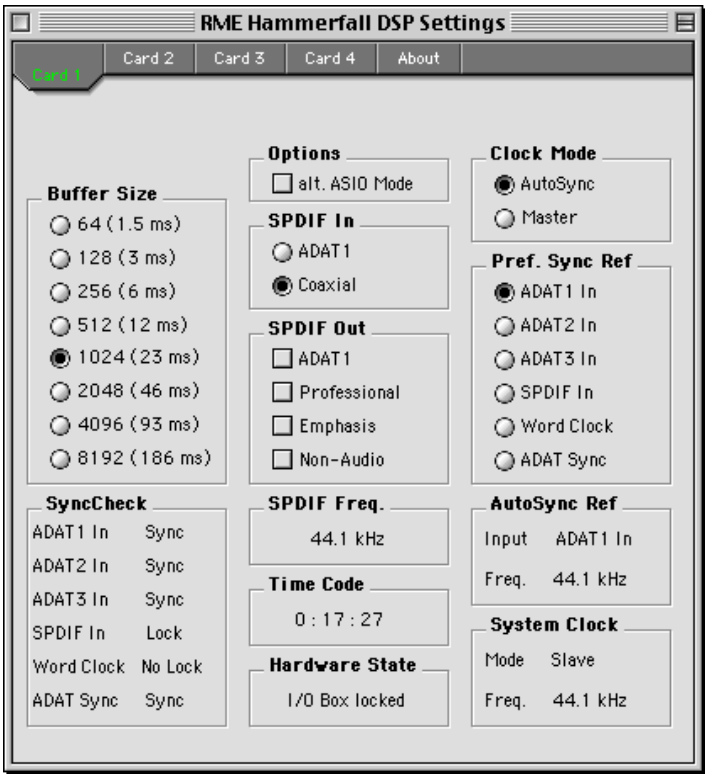
9. Configuring the HDSP 9652

9.1 General Information

Configuring the HDSP system is done using its own settings dialog, the program *Hammerfall DSP Settings*.

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

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



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

The status displays at the bottom of the dialog box give the user precise information about the current status of the system, and the status of all signals. 'SyncCheck' indicates whether there is a valid signal for each input ('Lock' or 'No Lock'), or if there is a valid *and* synchronous signal ('Sync'). The 'AutoSync Ref' display shows the input and frequency of the current sync source.

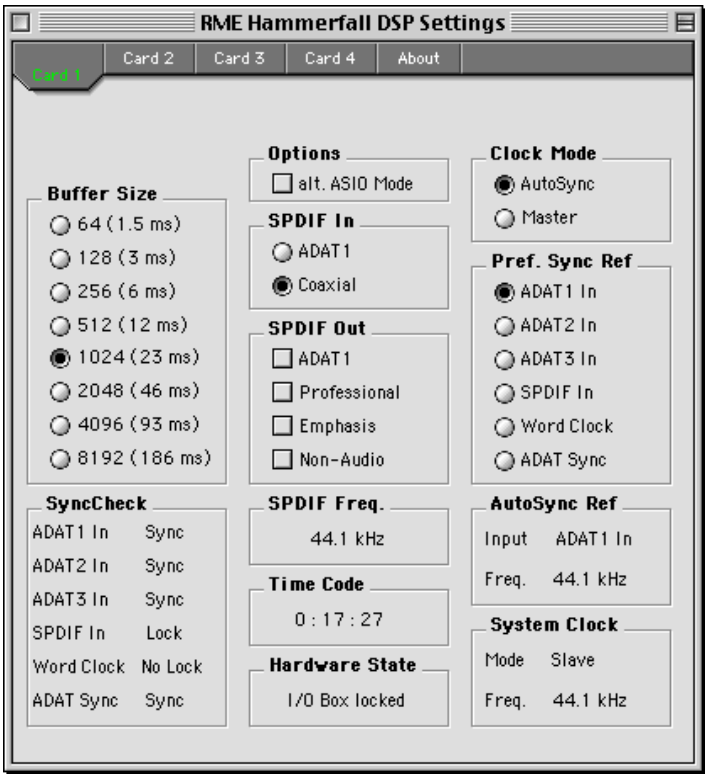
'Time Code' displays time information received from the I/O-box ADAT Sync port. This is convenient for checking whether the system is running in time with the transmitting device (e.g. ADAT).

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

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).

SPDIF In
Defines the input for the SPDIF signal. 'Coaxial' relates to the phono socket, 'ADAT1' to the optical input ADAT1.

SPDIF Out
The SPDIF output signal is constantly available at the phono plug. After selecting 'ADAT1' it is also routed to the optical output ADAT1. For further details about the settings ‘Professional’, ‘Emphasis’ and ‘Non-Audio’, please refer to chapter 12.



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

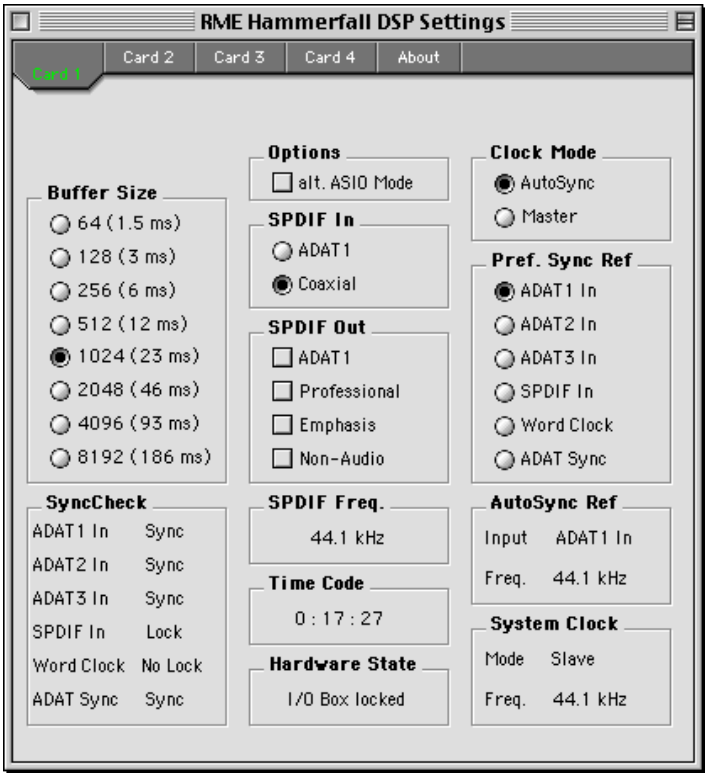
Pref. Sync Ref
Used to pre-select the desired clock source. If the selected source isn't available the card will change to the next available one. The currently used clock source and sample rate is displayed in the *AutoSyncRef* display.

The automatic clock selection checks and changes between the clock sources ADAT optical, SPDIF, word clock and ADAT Sync. The latter is recommended especially for sample-accurate transfers under ASIO 2.0.

System Clock
Shows the current clock state of the HDSP system. The system is either Master (using its own clock) or Slave (AutoSync Ref).

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 DSP’s intelligent clock control is very user-friendly, being able to switch between clock modes automatically. Selecting 'AutoSync' will activate this mode.



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

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

All the ADAT optical inputs in the Hammerfall DSP as well as the SPDIF input will work simultaneously. Because there is no input selector however, the HDSP 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 system 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 A*utoSync Ref* status field, together with the current sample frequency.

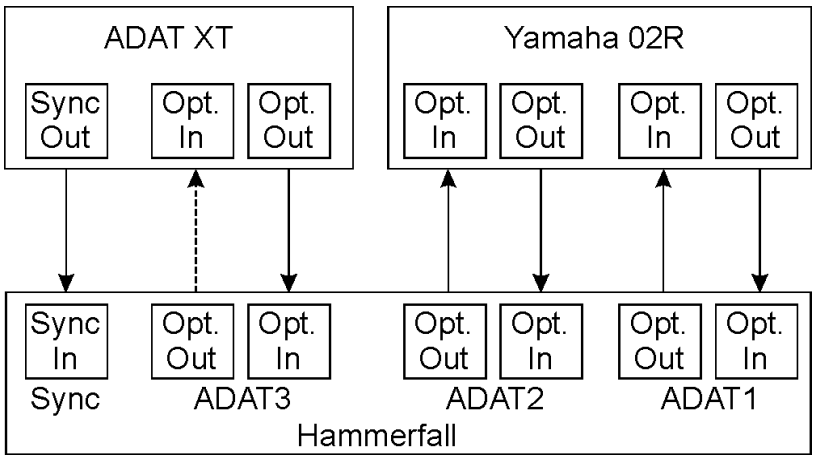
Via *Pref. Sync Ref* (preferred synchronization reference) a preferred input can be defined. As long as the card sees a valid signal there, this input will be designated as the sync source, otherwise the other inputs will be scanned in turn. If none of the inputs are receiving a valid signal, the card 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 SPDIF 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.

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 ‘AutoSync 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 Hammerfall DSP is set to AutoSync mode. As expected, SyncCheck shows that the ADAT1 and ADAT2 inputs are in sync (as they are driven by the same clock from the mixing desk), but shows ‘Lock’ instead of 'Sync' for the ADAT3 input. Because the ADAT recorder is not receiving any signals from HDSP 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 HDSP’s ADAT3 output. The Hammerfall DSP is already in sync with the mixing desk, so it will send an identical (synchronous) signal to ADAT3 out. The ADAT recorder will lock onto this, its output will also be in sync. The signal from the ADAT recorder is now fully in sync with the signals from the mixing desk.



Thanks to the its AutoSync technique and a lightning fast PLL, the HDSP 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 word clock is selected as input, 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 SPDIF 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. The Sync state of the SPDIF signal is shown by a blinking (locked) or constantly lit (Sync) input LED on the front of the Digiface.

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 ‘AutoSync 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.

Ideally all outputs of word clock delivering devices are designed with very low impedance, but all word clock inputs with high impedance, in order to not weaken the signal on the chain. But there are also negative examples, when the 75 Ohms are built into the device and cannot be switched off. In this case the network load is often 2 x 75 Ohms, and the user is forced to buy a special word clock distributor. Note that such a device is generally recommended for larger studios. 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 HDSP 9652's word clock input can be high-impedance or terminated internally, ensuring maximum flexibility. If termination is necessary (e.g. because HDSP 9652 is the last device in the chain), bring jumper X105 on the Expansion Board into position 75 Ohm.

In case the HDSP 9652 resides within a chain of devices receiving word clock, plug a T-adapter into its BNC input jack, and the cable supplying the word clock signal to one end of the adapter. Connect the free end to the next device in the chain via a further BNC cable. Remove jumper X105 from the Expansion Board. The last device in the chain should be terminated using another T-adapter and a 75 Ohm resistor (available as short BNC plug). Of course devices with internal termination do not need T-adaptor and terminator plug.

10.3 General Operation

The green ‘Lock’ LED at the Expansion Board 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, 'AutoSync Ref' will display 'Word'. This message has the same function as the green ‘Lock’ LED, but appears on the monitor, i.e. the user can check immediately whether a valid word clock signal is present and is currently being used.

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

11. Using more than one Hammerfall DSP

The current drivers support multiple interfaces and any combination of I/O-boxes (Multiface/Digiface/HDSP 9652). Please note that only one ADAT Sync can be used (of course). Additional all systems 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 DSP ignores the received header and creates a totally new one for the output signal.

Note that in record or monitor modes, set emphasis bits will disappear. Recordings originally done with emphasis should 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 DSP’s new output header is optimized for largest compatibility with other digital devices:

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

Professional AES/EBU equipment can be connected to the Hammerfall DSP thanks to the transformer-balanced coaxial outputs, and the ‘Professional’ format option with doubled output voltage. Output cables should have the same pinout as those used for input (see 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 SPDIF 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.

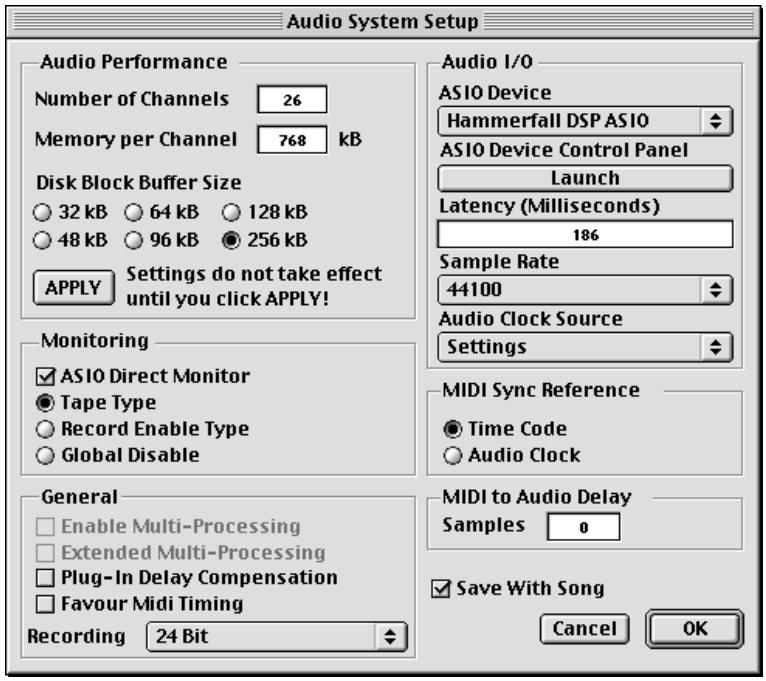
13. Operation under ASIO 2.0

13.1 General

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

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

Hammerfall DSP 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.

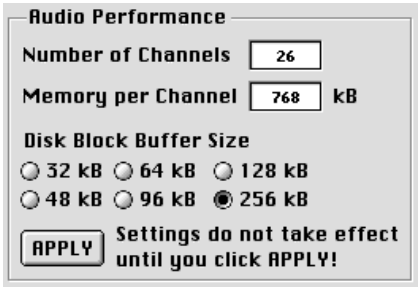


Hammerfall DSP supports 'ASIO Direct Monitoring' (ADM). Please note that at this time Cubase, Nuendo and Logic do not support ADM correctly. Bugfixes should be available soon.

For an operation at 88.2 and 96 kHz sample rate the device 'ASIO Hammerfall DSP 96 kHz' has to be chosen. When the sample frequency is set to 88.2 or 96 kHz, this driver operates all the ADAT optical inputs and outputs in Sample Split mode, so the number of available channels is reduced from 24 to 12.

13.2 Performance

The 'Audio Performance' settings are especially important. Firstly, the number of channels should be changed from 8 to 26 so that all the Hammerfall DSP's inputs can be accessed.



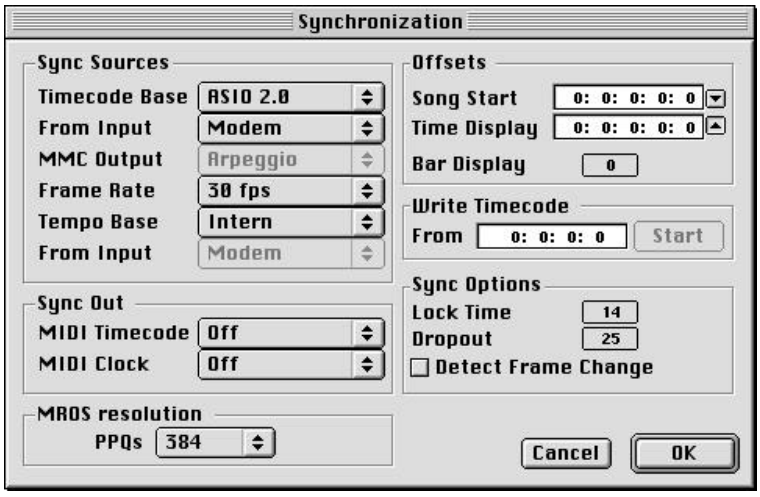
A very common problem is insufficient hard disk performance. If the first track is missing while recording multiple tracks, or the error message 'Audio: Record Error' appears, the disk sub-system is too slow i.e. it is unable to write the audio data to the disk quickly enough. The problem can almost 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. 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 10 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 DSP's Settings dialog determines the latency (in this case the delay) between the audio application and the HDSP 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.

13.3 Synchronization



To achieve sample-accuracy between the ADAT recorder and Hammerfall DSP while running Cubase, connect the ADAT sync output with the 9-pin D-type sync input of the HDSP. 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' 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'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
- Use the Clock Mode *ADAT Sync*

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.

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 DSP 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. TotalMix: Routing and Monitoring

The Hammerfall DSP system includes a powerful digital real-time mixer. RME's unique TotalMix technology allows for nearly unlimited mixing and routing with all inputs and playback channels simultaneously.

Here are some typical applications for TotalMix:

- setting up delay-free submixes (headphone mixes)
- unlimited routing of inputs and outputs (free utilisation, patchbay function)
- distributing signals to several outputs at a time
- simultaneous playback of different programs over only one stereo channel
- mixing of the input signal to the playback signal (complete ASIO Direct Monitoring)
- integration of external devices (effects etc). in real-time
- mixdown of three ADAT inputs to one (realizing two additional inputs)

On page 30 you'll find a block diagram of the TotalMix mixer of the Digiface. It can help to understand the basic signal flow and routing. It shows that the record signal always stays unaltered, but can be passed on as often as desired, even with different levels. The level meter of inputs and playback channels are connected pre-fader (due to the enormous routing capabilities). The level meters of the hardware's outputs are connected post-fader.

To call up the mixer start the program **Hammerfall DSP TotalMix**.

14.1 Elements of the Surface

The visible design of the mixer is mainly determined by the architecture of the HDSP system:

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

Every single channel has various elements:

Input and playback channels each have a mute and solo button.

Below each there is the panpot, realized as indicator bar (L/R) in order to save space.

In the window below this, the present level is displayed in RMS or Peak, being updated about every half a second. Overs are indicated here by an additional red dot.

Then comes the fader with a levelmeter. The meter shows both peak values (zero attack, 1 sample is enough for displaying full scale) by means of a yellow line and mathematically correct RMS values by means of a green bar. The RMS display has a relatively slow time constant, so that it shows the average loudness quite well.

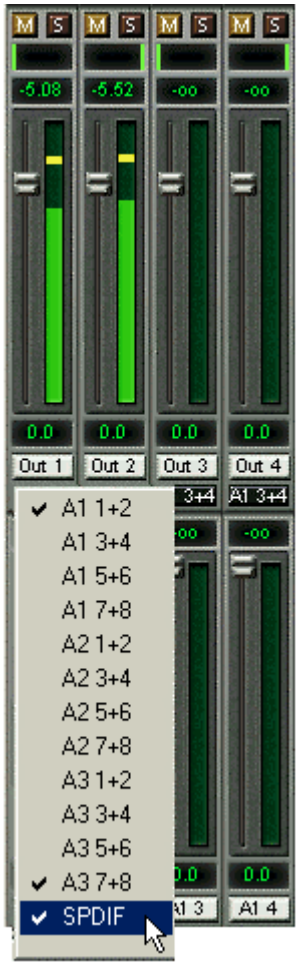
Below the fader, the current gain and panorama values are shown.

The white area shows the channel name, the black area shows the current routing target.

14.2 Tour de TotalMix

In the following chapters we will explain all functions of the surface step by step. Starting up TotalMix, the last settings are recalled automatically. When executing the application for the first time, a default file is loaded, sending all playback tracks 1:1 to the corresponding hardware outputs with 0 dB gain. The faders in the upper row are set to maximum attenuation (called m.a. in the following), so there is no monitoring of the input channels.

We will now create a small submix for the SPDIF output. Please start a multitrack playback and connect a DA-converter to the SPDIF output. In playback channel 1 (labeled 'Out 1'), click onto the routing window below the label. A list pops up, showing a checkmark in front of 'A1 1+2'. Click onto 'SPDIF'. The list disappears, the routing window no longer shows 'A1 1+2', but 'SPDIF'. Now move the fader with the mouse. As soon as the fader value is unequal m.a., the present state is being stored and routing is activated. Move the fader button to around 0 dB. The present gain value is displayed below the fader in green letters. In the lower row, on channels 25 and 26 (SP.L. and SP.R.), you can also see the level of what you are hearing. The level meter of the hardware output shows the outgoing level. Click into the area above the fader and drag the mouse in order to set the panorama, in this case the routing between channels 25 and 26. The present pan value is also being displayed below the fader.



Please carry out the same steps for 'Out 2' now, in order to route it to the analog output as well.

Often signals are stereo, i. e. a pair of two channels. It is therefore helpful to be able to make the routing settings for two channels at once. Press the Ctrl-key and click into the routing window of 'Out 3' with the key pressed. The routing list pops up with a checkmark at 'A1 3+4'. Click onto 'SPDIF'. Now, channel 4 has already been set to 'SPDIF' as well.

When you want to set the fader to exactly 0 dB, this can be difficult, depending on the mouse configuration. Move the fader close to the 0 position and now press the Shift-key. This activates the fine-mode, which stretches the mouse movements by a factor of 8. In this mode, a gain setting accurate to 0.1 dB is no problem at all.

Please set 'Out 4' to a gain of around -20 dB and the pan close to center. Now click onto the routing window. You'll now see two checkmarks, one at 'A1 3+4', the other one at 'SPDIF'. Click onto 'A3 7+8'. The window disappears, fader and panpot jump to their initial values, the signal can now be routed to the last ADAT channels. You can continue, until all entries have got a checkmark, i. e. you can send the signal to all outputs simultaneously. This is one of several differences to the Cubase mixer, which does not allow for multiple selections.

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

The checkmarks are un-checked by moving the fader to m.a. This setting deactivates the routing...why route if there is no level? Click onto 'A1 3+4' in the routing window, pull the fader down, open the routing window again - the checkmark is gone.

14.3 Submix View

Such a wide range of possibilities make it difficult to maintain the overview. Because practically all hardware outputs can be used for different submixes, as shown. And when opening the routing windows you might see an army of checkmarks, but you don't get an overwiev, i.e., how the signals come together and where. This problem is removed by the view mode 'Submix'. In this mode, all routing windows jump to the routing pair just being selected. So you can then see immediately, which channels, which fader and pan settings make a submix (for example 'SPDIF').

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

14.4 Mute and Solo

Mute works pre-fader, thus mutes all active routings of the channel. As soon as any Mute button is pressed, the Master Mute button lights up in the quick access area. It can switch all selected mutes off and on again. You can comfortably make mute groups to activate and deactivate this way.

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

14.5 Hotkeys

TotalMix knows only a few, but very effective key combinations, that make setting the mixer up considerably easier and faster. The Shift-key for the fine-mode for faders and panpots has already been mentioned. But the Ctrl-key can do far more than changing the routing pairwise:

- Clicking anywhere into the fader area with the Ctrl-key pressed, sets the fader to 0 dB, -6 dB for the hardware outputs.
- Clicking anywhere into the pan area with the Ctrl-key pressed, sets the panorama to <C> meaning 'Center'.

The faders can also be moved pairwise, corresponding to the basic stereo pairs. This can be achieved by pressing the Alt-key and is especially comfortable when setting the SPDIF and analogue output level. Even the Panoramas can be operated with Alt, from stereo through mono to inversed channels. At the same time, TotalMix also supports combinations of these keys. If you press Ctrl and Alt at the same time, clicking with the mouse makes the faders jump to 0 dB pairwise, and they can be set pairwise by Shift-Alt in fine-mode.

Also very useful: the faders have two mouse areas. The first area is the fader button, which can be grabbed at any place without changing the position. This avoids unwanted changes when clicking onto it. The second area is the whole fader setting area. Clicking into this area makes the fader jump to the mouse at once. If you want to set several faders to m.a. for instance, it is sufficient to click onto the lower end of the fader path. Which happens pairwise with the Alt-key pressed.

Using the hotkeys I, O and P the complete row each of Input, Playback and Output channels can be toggled between visible and invisible. Hotkey S switches Submix view on/off. Those four hotkeys have the same functionality as the buttons in the View section of the Quick Access Panel. The Level Meter Setup dialog can be opened by pressing the key L.

Further hotkeys are available to control the configuration of the Level Meter (see chapter 14.8):

Key 4 or 6: Display range 40 or 60 dB
Key E or R: Numerical display showing Peak or RMS
Key 0 or 3: RMS display absolute or relative to 0 dBFS

14.6 The Quick Access Panel

This section includes additional options, further improving the handling of TotalMix. The Master button for Mute and Solo has already been described, they allow for group-based working with these functions.

In the **View** section the single rows can be made visible or invisible. If the inputs are not needed for a pristine playback mix, the whole upper row falls out of the picture after a click on the input button. If the hardware outputs don't interest you either, the surface can thus be reduced to the playback channels to save space. All combinations are possible.

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

The mixer can also be made smaller horizontally, and, scrolled. TotalMix can be made substantially smaller and space-saving on the desktop/screen, if you have to have to monitor or set only a few channels or level meters.

The **Presets** are one of the mightiest and most useful features of TotalMix. Behind the eight buttons, eight files are hidden (see next chapter). These contain the complete mixer state. Just try it: all faders and other settings follow the changing of preset(s) in real-time, just by a single mouse click. The Save button allows for storing the present settings in the present preset. You can change back and forth between a signal distribution, complete input monitoring, a stereo and mono mix, and various submixes without any problem.

Also here, RME's love for details can be seen. If any parameter is being altered after loading a preset (e. g. moving a fader), the preset display flashes in order to announce that something was changed, still showing, which state the present mix is based on.

If no preset button is lit, another preset had been loaded via the File menu and 'Open file'. Mixer settings can of course be saved the usual way, and with long file names.

Up to three Hammerfall DSP systems can be used simultaneously. The **Card** buttons switch between the systems. Systems, because card 1 can be a Digiface, but card 2 can also be a Multiface.

The number of ADAT channels is reduced to half automatically when chosing double speed operation (88.2 or 96 kHz). The display is adjusted accordingly, but the fader settings remain stored.

14.7 Presets

During the driver installation 8 factory presets are copied to the preferences folder (inside the folder 'Hammerfall DSP'). Those files are named **preset1.mix** to **preset8.mix**, and will be used when clicking on the 8 Preset buttons in the Quick Access Panel.

But TotalMix will read those files only at first usage. As soon as one of the Presets is saved, TotalMix writes a new file and adds the number of the currently used system (Card 1, 2 or 3). The files **preset1.mix** thus changes to **preset11.mix**, if Card 1 was active. This method offers two major advantages:

- Presets modified by the user will not be overwritten when reinstalling or updating the driver
- The factory presets remain unchanged, and can be reloaded anytime using the menu, Files/Open

The 8 factory presets offer not only a useful functionality for TotalMix, but also a pretty good base to modify them to your personal needs.

Preset1.mix
Description: All channels routed 1:1, playback monitoring via headphone out

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

Note: This preset is *Default*, offering the standard functionality of a I/O-card.

Preset2.mix
Description: All channels routed 1:1, input and playback monitoring via Phones. As Preset 1, plus submix of all inputs (0 dB) on Phones.

Preset3.mix
Description: All channels 1:1, input and playback monitoring via Phones and outputs. As Preset 2, but all inputs set to 0 dB (1:1 pass through).

Preset4.mix
Description: All channels 1:1, playback monitoring via Phones and outputs. As Preset 3, but all inputs muted.

Preset5.mix
Description: All faders m.a. As Preset 1, but all outputs m.a.

Preset6.mix
Description: Submix on SPDIF at -6 dB. As Preset 1, plus submix of all playbacks on SPDIF. View Submix SPDIF active.

Preset7.mix
Description: Submix on SPDIF at -6 dB. As Preset 6, but submix of all inputs and outputs on SPDIF. View Submix SPDIF active.

Preset8.mix
Description: Panic. As Preset 4, but also playback muted (no output signal).

14.8 Level Meter

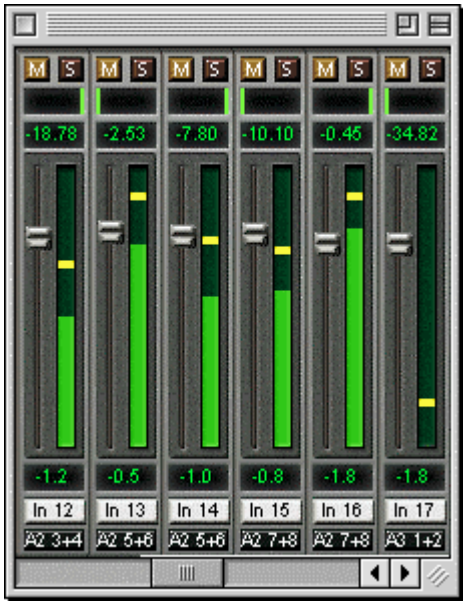
Having set a new standard with the level meters of DIGICheck, Hammerfall DSP goes even further: The calculation of the Peak, RMS and Over is realized in hardware, in order to be capable of using them independent of the software in use, and to significantly reduce the CPU load.

The level meters integrated in TotalMix - considering their size - cannot be compared with the *HDSP Meter Bridge* (available later). Nevertheless they already include many useful functions.

Peak and RMS is displayed for every channel. 'Level Meter Setup' (Menu Options) or direct keyboard entry (*hotkeys*) makes various options available:

- Display range 40 or 60 dB (*hotkey 4 or 6*)
- Release time of the Peak display (Fast/Medium/Slow)
- Numerical display selectable either Peak or RMS (*Hotkey E or R*)
- Number of consecutive samples for Overload display (1 to 15)
- RMS display absolute or relative to 0 dBFS (*Hotkey 3 or 0*)

The latter is a point often overlooked, but nonetheless important. RMS shows 3 dB less for sine signals. This is mathematically correct, but not very reasonable for a level meter. Therefore, we had corrected DIGICheck's RMS display by 3 dB, a full scale sine signal shows both 0 dBFS Peak and RMS. This setting also yields directly readable signal-to-noise values, while other applications (like WaveLab) will show a value 3 dB better than actual (because the reference is not 0 dB, but -3 dB).



The value displayed in the text field is independent of the setting 40/60 dB, it represents the full 24 bit range of the RMS measurement, thus making possible a SNR measurement 'RMS unweighted', which you would otherwise need extremely expensive measurement devices for. An ADI-8 DS connected to the Digiface will therefore show around -113 dB on all 8 channels.

This level display will constantly bring the reduced dynamic range of your equipment, maybe of the whole studio, in front of your eyes. Nice to have everything 24 bit - but still noise and hum everywhere in the range around -90 dB or worse... sorry, but this is hard reality. The up-side about it is that TotalMix allows for constantly monitoring the signal quality without effort. Thus it can be a valuable tool for sound optimization and error removal in the studio.

Measuring SNR (Signal to Noise) requires to press R (for RMS) and 0 (for referring to 0 dBFS, a full scale signal). The text display will then show the same value as an expensive measurement system, when measuring 'RMS unweighted'.

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

15. Hotline - Troubleshooting

15.1 General

The newest information can always be found on our website www.rme-audio.com, section MacOS, Hammerfall DSP Support.

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 25 and 26).

Playback works, but record doesn’t:

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

Crackle during record or playback:

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

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.

Digital Performer and Logic crash when used at 96 kHz

- This problem (missing reset for changed number of channels in Double Speed operation) is solved by the special ASIO driver 'Hammerfall DSP ASIO 96 kHz'. Select this driver within the corresponding program when working in 88.2 and 96 kHz.

15.2 Installation

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

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

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

- Check whether the Hammerfall DSP has been selected as current ASIO device.

The following symptoms are typical for PCI related problems:

- When booting the control panels are displayed too big, or 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).

16. Software and Hardware Compatibility

Hammerfall DSP is fully compatible with PCI bus version 2.1.

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, Max/MSP from Cycling '74 and Super Collider.

As far as we are aware, the Hammerfall DSP digital inputs and outputs are fully compatible with all devices with SPDIF or AES/EBU interfaces.

17. Accessories

RME offers several optional components, further increasing the flexibility and usability of the HDSP system. Additionally parts of the HDSP system, like the brealout cables, are available seperately.

OK05	Optical cable, Toslink, 0.5 m (1.5 ft)
OK1	Optical cable, Toslink, 1 m (3.3 ft)
OK2	Optical cable, Toslink, 2 m (6.6 ft)
OK3	Optical cable, Toslink, 3 m (9.9 ft)
OK5	Optical cable, Toslink, 5 m (16.4 ft)
OK10	Optical cable, Toslink, 10 m (32.8 ft)

Standard lightpipe with TOSLINK connectors, RME approved quality.

BO9652	Breakout cable SPDIF
BOHDSP9652	Breakout cable MIDI
VKHDSP9652	Internal flat cable 14-core

18. TECH INFO

Not all information to and around our products fit in a manual. Therefore RME offers a lot more and detailed information in the **Tech Infos**. The very latest Tech Infos can be found on our website, section News & Infos, or the directory **\rmeaudio.web\techinfo** on the RME Driver CD. These are some of the currently available *Tech Infos*:

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.

Configuring Logic, Samplitude and Cubase for the DIGI32/96 series
Configuring Cakewalk and SAWPlus32 for the DIGI32/96 series
Step by step instructions.

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).

HDSP System: Notebook Basics - Notebook Hardware
HDSP System: Notebook Basics - The Audio Notebook in Practice
HDSP System: Notebook Basics - Background Knowledge and Tuning
HDSP System: Notebook Tests - Compatibility and Performance
Many background information on laptops. Tests of notebooks

HDSP System: TotalMix - Hardware and Technology
HDSP System: TotalMix - Software, features, operation
The digital mixer of the Hammerfall DSP in theory and practise

19. Warranty

Each individual Hammerfall DSP undergoes comprehensive quality control and a complete test in a PC environment at RME before shipping. This may cause very slight signs of wear (if it 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 DSP. 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:

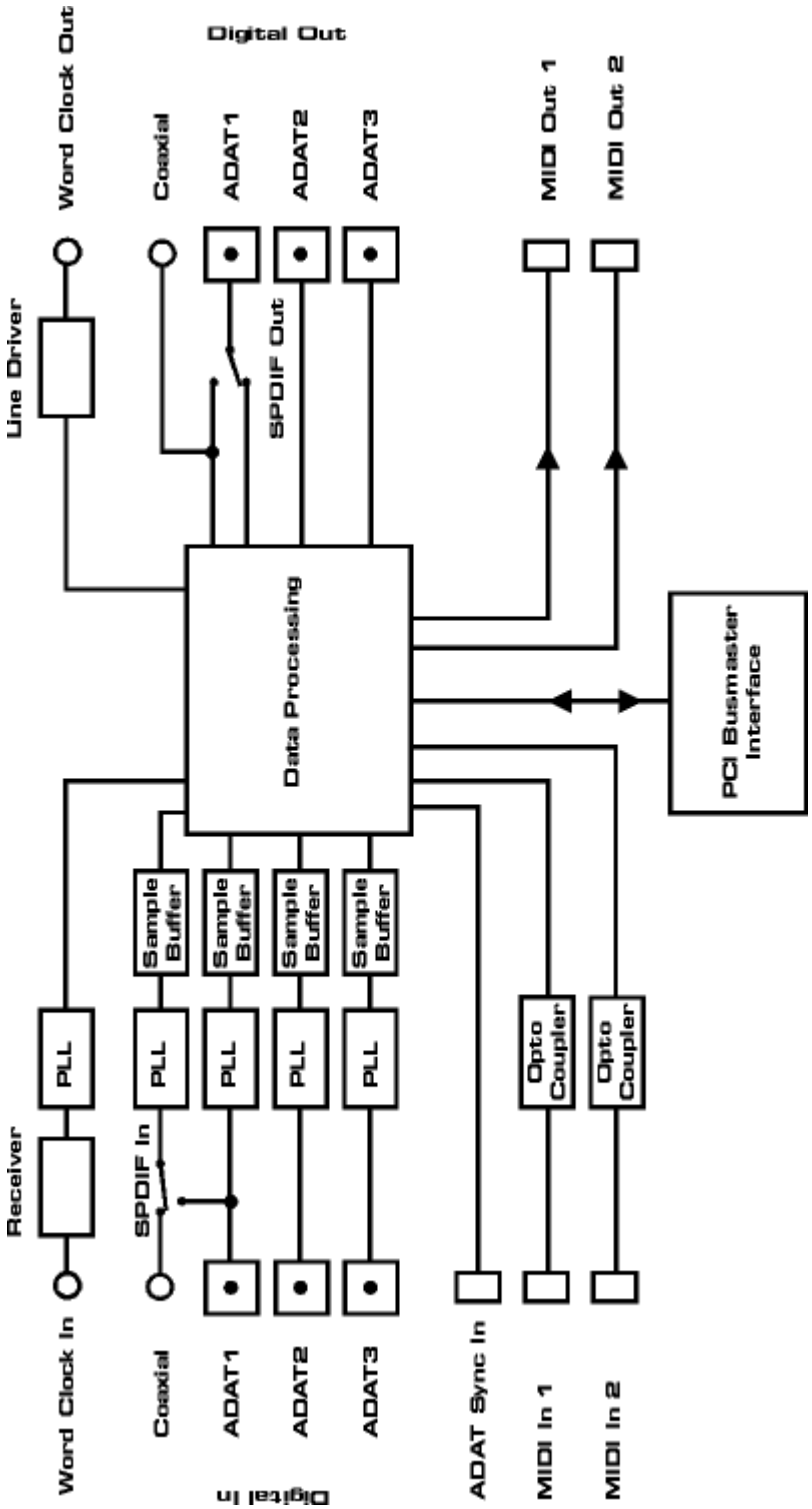
http://www.rme-audio.com

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.

Trademarks

21. Diagrams

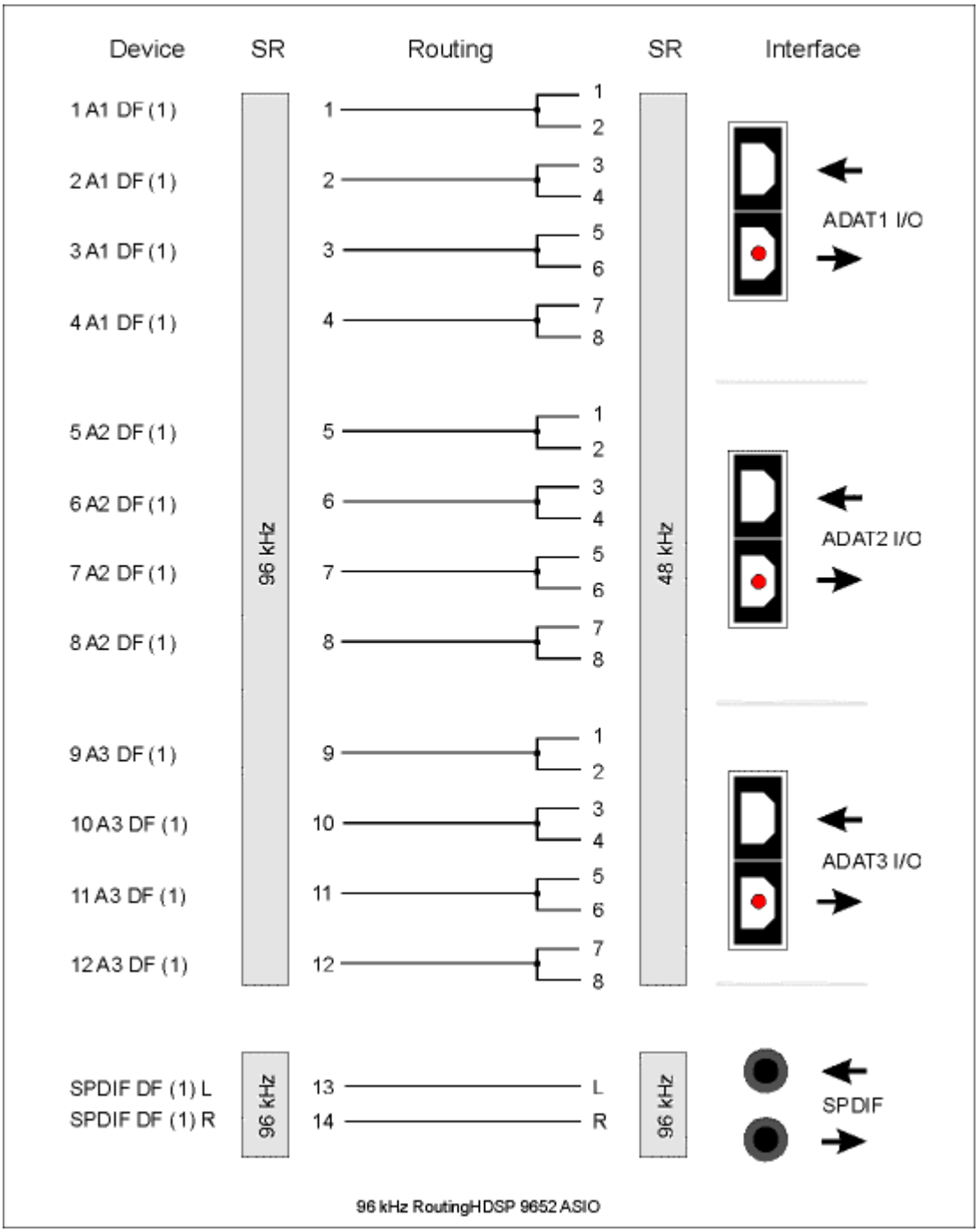
21.1 Block Diagram HDSP 9652



21.2 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.

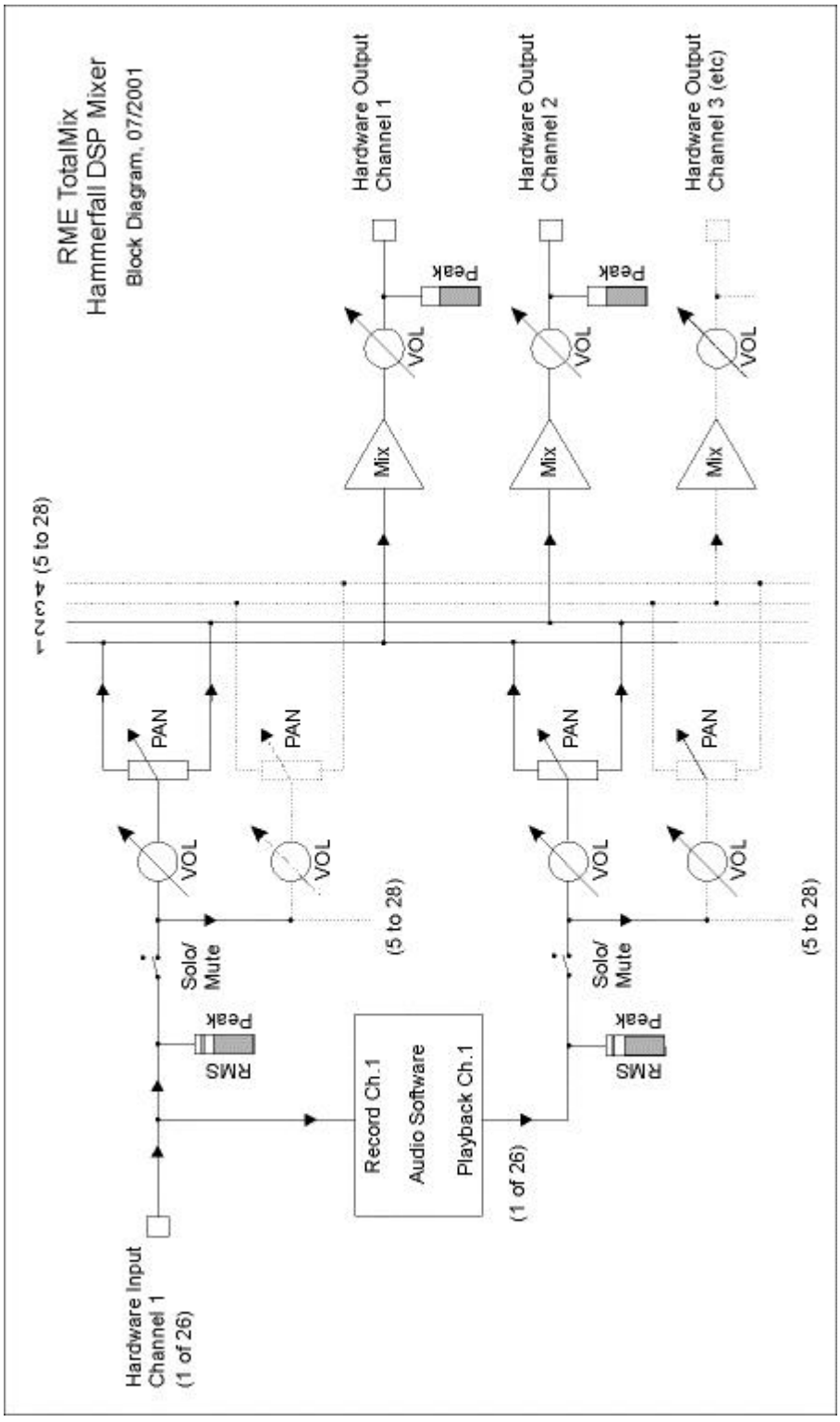
Device: The device name in the audio application SR: Sample Rate
Device name code: Channel in ASIO host, ADAT interface, Digiface, card number



21.3 Block diagram TotalMix

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

- The input signal of the hardware (ADAT/SPDIF In) is always directly fed through to the recording software. At the same time it can be routed to all 26 hardware outputs (ADAT/ SPDIF), even to all of them simultaneously.



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.