User's Guide

RME
INTELLIGENT AUDIO SOLUTIONS

Hammerfall® DSP System

HDSP 9652

DVD
24 Bit / 96 kHz ✓ *ready*

TotalMix™

◄ASIO►

SyncAlign®   ZLM®   SyncCheck®

adat®
SYNC

adat®
OPTICAL

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. Installation is simple, even for the inexperienced user, thanks to the latest Plug and Play technology and full interrupt-sharing. The numerous unique features and well thought-out configuration 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 Mac OS. An ALSA driver for Linux is also available (see chapter 7.6).

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-conductor)
- 2 optical cable (TOSLINK), 2 m (6.6 ft)

3. System Requirements

- Windows 98/SE/ME, Windows 2000/XP, Linux, Mac OS 9.x or X
- PCI Interface: a free PCI rev. 2.1 Busmaster slot

Note: Examples and detailed descriptions of suitable audio desktop systems can be found in the Tech Info *RME Reference PCs: Hardware recommendations*.

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: ADAT In, SPDIF In, and all outputs can be used simultaneously
- 8 available buffer sizes/latencies: 1.5 / 3 / 6 / 12 / 23 / 46 / 93 / 186 ms
- Sample Split technology for 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 0.8 V (consumer mode, phono) or 2.3 V (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

ASIO:
* 24 or 32 bit, 4 byte (stereo 8 byte)

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

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

Channel Interleave operation is limited to single speed operation and 16 bit.

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. Driver Installation

7.1 Windows 98/SE/ME

After the PCI card has been installed correctly (see 6. Hardware Installation), and the computer has been switched on, Windows will recognize the new hardware component and start its 'Add New Hardware Wizard'. Insert the RME Driver CD into your CD-ROM drive, and follow further instructions which appear on your computer screen. The driver files are located in the directory **\HDSP_w98** on the RME Driver CD.

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

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

HDSP 9652 can be easily configured using the HDSP system's Settings dialog (see section 9.1)

7.2 Windows 2000/XP

After the interface has been installed correctly, connected to the Digiface (see 6. Hardware Installation), and the computer has been switched on, Windows will recognize the new hardware component and start its 'Hardware Wizard'. Insert the RME Driver CD into your CD-ROM drive, and follow further instructions which appear on your computer screen. The driver files are located in the directory **\HDSP_w2k** on the RME Driver CD.

Windows will install the Hammerfall DSP System driver, and will register the card in the system as a new audio device. After a reboot the HDSP 9652 is ready for use.

HDSP 9652 can be easily configured using the HDSP's Settings dialog (see section 9.1).

In case the warning messages 'Digital signature not found', 'Do not install driver', 'not certified driver' or similar come up: Don't listen to Microsoft, listen to us and continue with the installation.

☞ *In case the Hardware Wizard does not show up automatically after installation of the card, do not attempt to install the drivers manually! An installation of drivers for non-recognized hardware will cause a blue screen when booting Windows!*

7.3 Driver Update

RME's driver updates often include a new hdsp32.inf file. Also the revision number of the hardware might change (after a flash update). To prevent Windows 2000/XP from using an old hdsp32.inf, or to copy some of the old driver files, be sure NOT to let Windows search for the driver! Instead tell Windows what to do.

Under Control Panel /System /Device Manager /Sound, Video and Game Controllers /RME Hammerfall DSP/Properties /Driver you'll find the 'Update Driver' button. Select '**Install from a list or specific location (advanced)**', click '**Next**', select '**Don't search I will choose the driver to install**', click '**Next**', then '**Have Disk**'. Now point to the driver update's directory.

7.4 Flash Update

The Flash Update Tool updates the HDSP 9652's hardware to the latest version. It requires an already installed driver.

Start the program fut_xxx_xxx.exe. The Flash Update Tool displays the current revision of the HDSP interface, and whether it needs an update or not. If so, then simply press the 'Update' button. A progress bar shows how several actions are performed. When the flash update process is finished, 'Success' will be displayed.

If more than one interface cards are installed, all cards can be flashed by changing to the next tab and repeating the process.

After the update the PCI card needs to be resettet. This is done by powering down and shutting off the PC. A warm boot is not enough.

When the update fails (status: failure), the card's second BIOS will be used from the next cold boot on (Secure BIOS Technology). Therefore the card stays fully functional. The flash process should then be tried again on a different computer.

Note for Windows 2000/XP users: Because of the changed hardware revision, Windows 2000/XP will start the hardware assistant and wants to install new drivers. Do NOT let Windows search for new drivers, but follow the instructions given in chapter 7.3 and manually perform a driver update.

7.5 Deinstalling the Drivers

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

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

7.6 Linux/Unix

An ALSA driver for Linux is available. TotalMix has been ported to Linux too. 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.

ADAT2   ADAT1   SPDIF I/O
Out   In   Out   In   ADAT Sync In
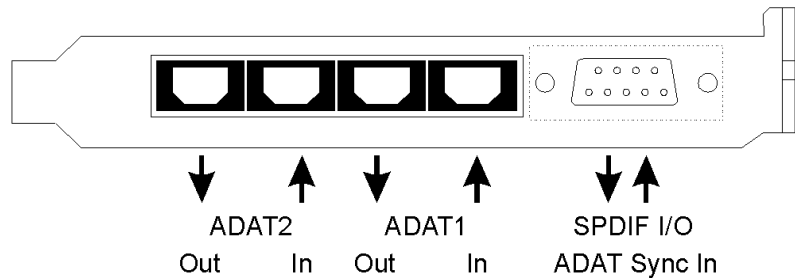
 The ADAT1 I/O next to the D-type socket can also be used for optical SPDIF, if this mode is selected in the Settings dialog.

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.
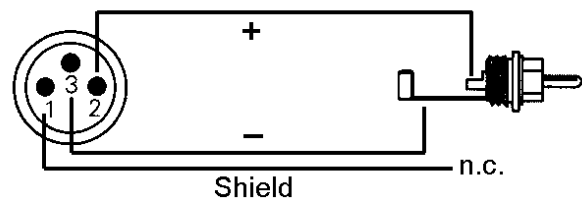
The ground-free design using transformers for digital inputs and outputs enables trouble-free connection to all devices, and perfect hum rejection.

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.

MIDI   ADAT3   Word Clock   Lock LED
Out/In   Out   In   Out   In
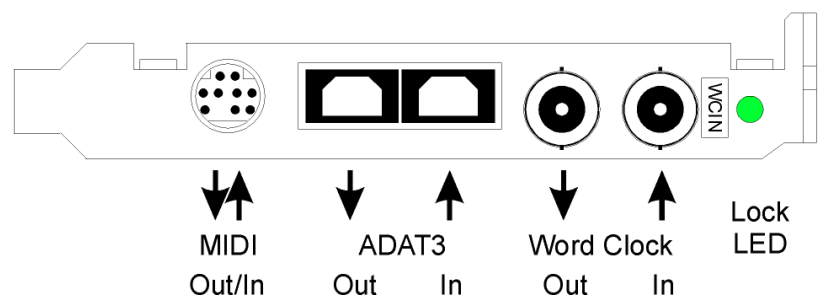
The included breakout cable is connected to the 9-pin Mini-DIN connector and realizes two MIDI inputs and outputs.

8.2 Internal Connections

AEB1 IN / CD IN
This internal digital input can be used with both SPDIF and ADAT.

SPDIF:
- Connection to an internal CD-ROM drive with digital audio output. Allows for a direct transfer of digital audio data within the computer.
- Connection to SYNC OUT of another card. This internal SPDIF connection can be used to synchronize multiple cards with sample accuracy, and without the need for an external connection. The card of which SYNC OUT is used will be master, the SYNC IN one will be slave. *SPDIF In / Internal* has to be selected in the Settings dialog. Additionally *Pref. Sync Ref* has to be set to *SPDIF In* for this internal connection to work properly. Please note that the external SPDIF- or AES input can no longer be used then.

ADAT:
- Connection to an AEB4-I or AEB8-I. When using these Expansion Boards ST7 must also be connected to the Expansion Board. The highest sample rate is 48 kHz. Select *AEB / ADAT Int.* in the Settings dialog. In this mode, the optical input can still be used as optical SPDIF input.
- Connection to a TEB (TDIF Expansion Board). The highest sample rate is 96 kHz, the 4-channel *Dual Line Mode (equals SMUX)* is automatically activated in double speed mode. Select *AEB / ADAT Int.* in the Settings dialog.

AEB2 IN
Jumper ST5 allows to connect a second AEBx-I. Remove the jumper (factory default 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.

ADAT 1 OUT
This internal ADAT output carries the same audio data as the optical output ADAT1 in ADAT mode. Connecting an AEB4-O or AEB8-O, the highest sample rate is 48 kHz. Connecting a TEB the highest sample rate is 96 kHz, the 4-channel *Dual Line Mode (equals SMUX)* is automatically activated in double speed mode. The internal ADAT output stays active, even when the optical output is switched into *SPDIF* operation. Please note the label GND for correct polarity.

ADAT 2 OUT
This internal ADAT output carries the same audio data as the optical output ADAT2. See ADAT 1 OUT for details. Both ports can be used to operate one AEBx-O each, for a maximum of 16 analog outputs.

SYNC OUT
This internal SPDIF output carries the same audio data as the external phono or XLR output. It can be used to synchronize multiple cards, see above. Please note the label GND for correct polarity.

X6
14-pin connector for the Expansion Board connection via flat ribbon cable.

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 Playback

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

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

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

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

The screenshot to the right shows a typical configuration dialog as displayed by a (stereo) wave editor. After selecting a device, audio data is sent either to SPDIF or to the ADAT ports, depending on which has been selected as playback device.

Increasing the number and/or size of audio buffers may prevent the audio signal from breaking up, but also increases latency i.e. output is delayed.
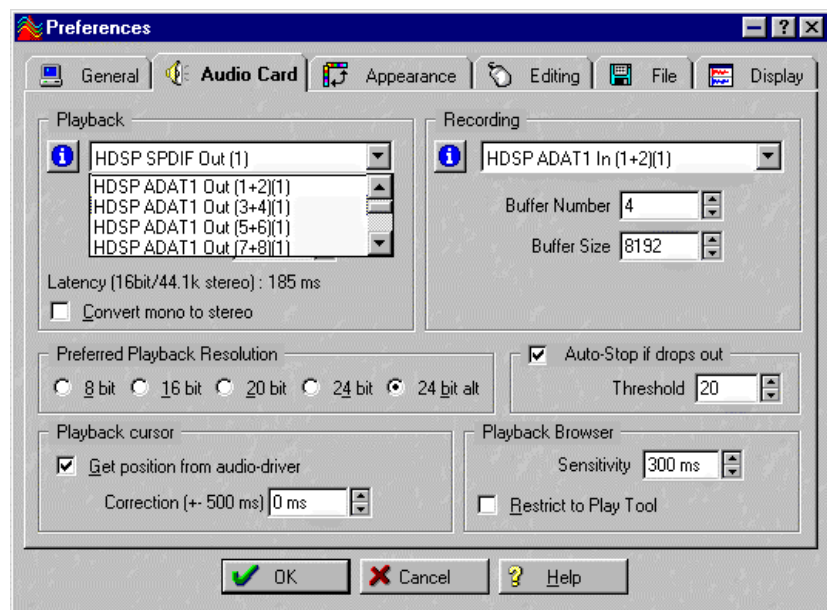
For synchronized playback of audio and MIDI (or similar), be sure to activate the checkbox 'Get position from audio driver'.

The HDSP system's ADAT optical interface allows sample rates of up to 96 kHz using a standard ADAT recorder. Single-channel data at this frequency requires two ADAT channels, achieved using the 'Sample Split' technique. This reduces the number of available ADAT channels from 24 to 12. Under Windows MME, channels are routed to ADAT devices in double-speed mode as follows:

- Only stereo pairs (1+2) and (3+4) of each ADAT port are available
- Channel 1 is routed to channels 1 and 2, channel 2 is routed to 3 and 4 etc.

Please refer to the diagram 'ADAT Track Routing, MME 96 kHz', section 24. Routing for record and playback is identical.

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

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

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

This 'SPDIF' signal sounds like chopped noise at highest level. Therefore the HDSP 9652 automatically activates the non-audio bit within the digital data stream, to prevent most SPDIF receivers from accepting the signal, and to prevent any attached equipment from being damaged.

Multichannel
PowerDVD can also operate as software decoder, sending a DVD's multichannel data stream directly to any of the outputs of the HDSP 9652. Supported are all modes, from 2 to 8 channels, at 16 bit resolution and 48 kHz sample rate.

For this to work an output wave device of the HDSP 9652 has to be selected in 'Control Panel/Sounds and Multimedia/Audio'. Also check 'use preferred device only'. PowerDVD's audio properties now lists several multichannel modes. If one of these is selected, PowerDVD sends the decoded analog multichannel data to the HDSP.

The device selected as *Preferred Playback Device* defines the first playback channel. Choosing ADAT1 3/4 and 6-channel mode, playback will happen on channels 3 to 8. Choosing ADAT3 5/6 and 6-channel mode, even the SPDIF output will be used for the last (highest) channels.

The available modes depend on the number of channels available above the chosen device! When selecting ADAT3 7/8, the 6-channel mode will not be available.

The channel assignment using PowerDVD is:

1 (first chosen playback channel) - Left
2 - Right
3 - Center
4 - LFE (Low Frequency Effects)
5 - SR (Surround Right)
6 - SL (Surround Left)

Note 1: Setting the card to be used as system playback device is against common sense, as professional cards are not specialized to play back system sounds, and shouldn't be disturbed by system events. To prevent this, be sure to re-assign this setting after usage, or to disable any system sounds (tab Sounds, scheme 'No audio').

Note 2: The DVD player will be synced backwards from the HDSP card. This means when using AutoSync and/or word clock, the playback speed and pitch follows the incoming clock signal.

8.5 Low Latency under MME (Buffer Size Adjustment)

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

In the HDSP's Settings dialog the MME buffersize (in fact the DMA buffer size) is set with the same buttons as the ASIO buffer size. Our test computers allow to use settings down to 64 samples without clicks. Please note that this setting only defines the buffer size of the hardware. The true and effective latency is configured within the MME application!

☞ *Attention: the DMA buffers must not be larger than the application's buffers. This case can happen unnoticed when using ASIO and MME at the same time (multi-client) and setting ASIO to 186 ms, while the buffers in the MME application are still set for a lower latency. Playback will be stuttering and audio will be distorted.*

Example: when you set the Hammerfall to 512 you can't use 128 in any program. But setting DMA to 128 allows to use 128 and all higher values within the software.

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

8.6 Multi-Client Operation

RME audio cards support multi-client operation. This means several programs can be used at the same time. Also all formats, like ASIO, MME and GSIF can be used simultaneously. The use of multi-client operation requires to follow two simple rules:

- *Multi-client operation requires identical sample rates!*

It is not possible to use one software with 44.1 kHz and the other with 48 kHz.

- *Different software can not use the same channels at the same time.*

If for example Cubase uses channels 1/2 (default in Cubase, Master bus), this playback pair can't be used in Gigasampler/Studio (GSIF) nor under MME (WaveLab etc) anymore (the inputs can be used at the same time). This is no limitation at all, because TotalMix allows any output routing, and with this a playback of multiple software on the same hardware outputs.

ASIO Multi-client
RME audio cards support ASIO multi-client operation. It is possible to use more than one ASIO software at the same time. Again the sample rate has to be identical, and each software has to use its own playback channels. The inputs can be used simultaneously.

An exception is our sophisticated tool *DIGICheck*. It operates like an ASIO host, using a special technique to access playback channels already occupied. Therefore DIGICheck is able to perform an analyzis and display of playback data from any software, no matter which format the software uses.

8.7 Recording

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

To take this into account, RME have added two unique features to the HDSP 9652: a comprehensive I/O signal status display (showing sample frequency, lock and sync status) in the Settings dialog, and the protective *Check Input* function.

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

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

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

The screenshot to the right shows a typical dialog used for changing basic parameters such as sample frequency and resolution in an audio application.

Any bit resolution can be selected, providing it is supported by both the audio hardware and the software. Even if the input signal is 24 bit, the application can still be set to record at 16-bit resolution. The lower 8 bits (and therefore any signals about 96dB below maximum level) are lost entirely. On the other hand, there is nothing to gain from recording a 16-bit signal at 24-bit resolution - this would only waste precious space on the hard disk.
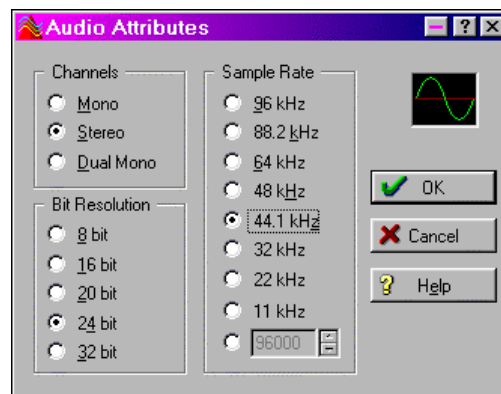
It often makes sense to monitor the input signal or send it directly to the output. This can be done at zero latency using **TotalMix** (see chapter 15).

TotalMix also includes a useful automatic real-time monitor function, see chapter 15.8 for details. Activating record in the application causes the input signal to be routed according to the current mixer settings.

Currently two solutions exist which enable an automated control of real-time monitoring. ZLM (Zero Latency Monitoring) allows monitoring in Punch I/O mode - with this the system behaves like a tape machine. This method has been implemented in all versions of Samplitude (by SEK'D), and can be activated using the global track option 'Hardware monitoring during Punch'.

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

9. Configuring the HDSP 9652
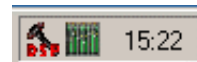
9.1 General Information

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

- by clicking on the hammer icon in the Taskbar's system tray
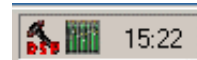
The mixer of the Hammerfall DSP System (TotalMix) can be opened in two different ways:

- by clicking on the mixer icon in the Taskbar's system tray

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 ADAT Sync port. This is convenient for checking whether the system is running in time with the transmitting device (e.g. ADAT).
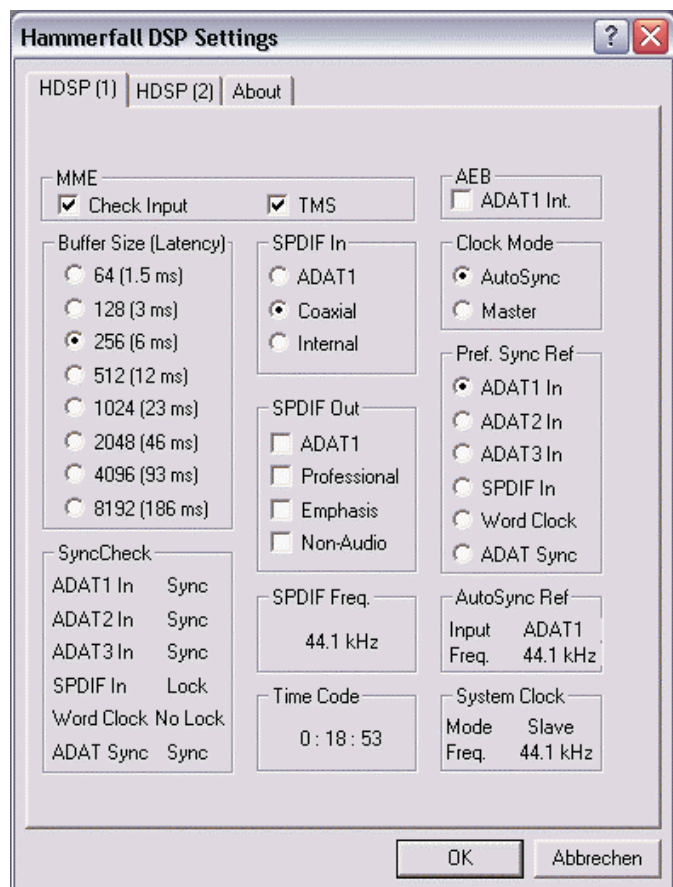
MME

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

TMS activates the transmission of Channel Status data and Track Marker information of the SPDIF input.

Buffer Size

The setting *Buffer Size* determines the latency between incoming and outgoing ASIO and GSIF data, as well as affecting system stability (see chapter 13). Under Windows MME this setting determines the DMA buffer size (see chapter 8.5).

SyncCheck

'SyncCheck' indicates for ADAT, ADAT Sync, SPDIF and word clock input whether there is a valid signal ('Lock' or 'No Lock'), or a valid *and* synchronous signal ('Sync'). The 'AutoSync Ref' display shows the input and frequency of the current sync source.

SPDIF In

Defines the input for the SPDIF signal. 'Coaxial' relates to the phono socket, 'ADAT1' to the optical TOSLINK 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).
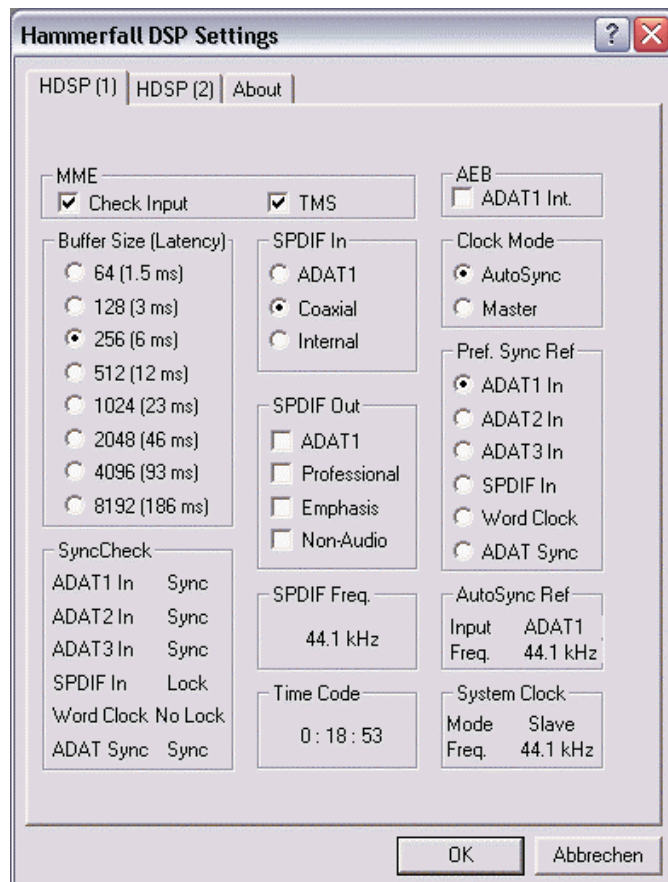
9.2 Clock Modes - Synchronization

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 'Master') to a clock generated from the input signal (AutoSync Ref displays 'Slave'). This allows on-the-fly recording, even during playback, without having to synchronize the card to the input signal first. It also allows immediate playback at any sample rate without having to reconfigure the card.

AutoSync guarantees that normal record and record-while-play will always work correctly. In certain cases however, e.g. when the inputs and outputs of a DAT machine are connected directly to the Hammerfall DSP, AutoSync may cause 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 *AutoSync Ref* status field, together with its 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 AutoSync 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.
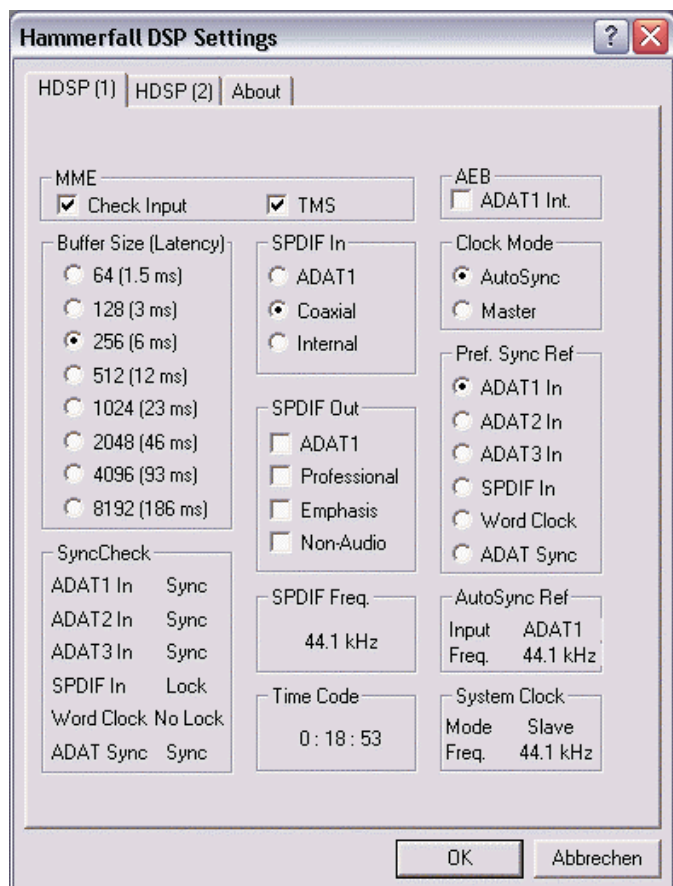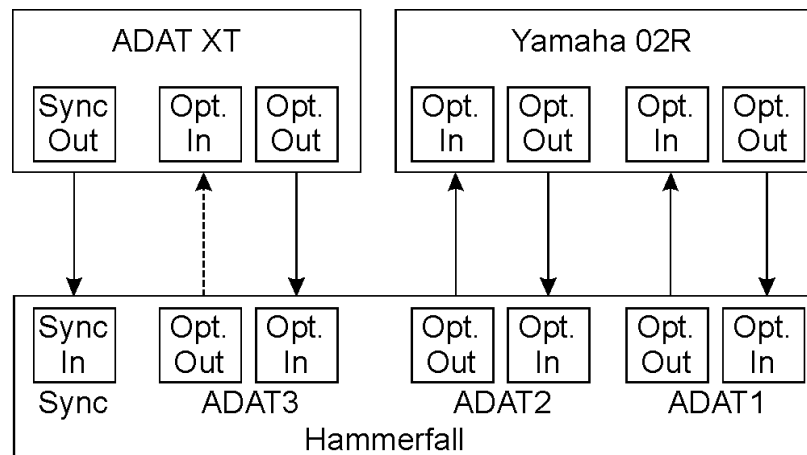
If several digital devices are to be used simultaneously in a system, they not only have to operate with the same sample frequency but also be synchronous with each other. This is why digital systems always need a single device defined as 'master', which sends the same clock signal to all the other ('slave') 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 digital clock source inputs. The 'AutoSync Ref' display shows the current sync source and the measured 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 its AutoSync technique and lightning fast PLLs, the HDSP is not only capable of handling standard frequencies, but also any sample rate between 25 and 105 kHz. Even the word clock input, which most users will use in varispeed operation, allows any frequency between 25 kHz and 103 kHz.

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/HDSP9632). 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 adapted 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.
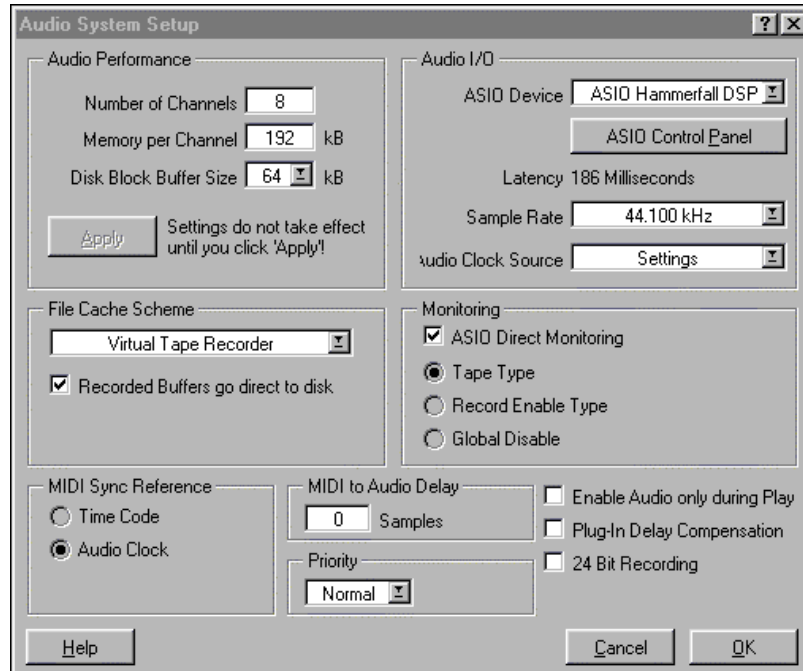
Hammerfall DSP supports 'ASIO Direct Monitoring' (ADM).

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 24 to 12.

13.2 Known Problems

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

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

Another common source of trouble is incorrect synchronization. ASIO does not support asynchronous operation, which means that the input and output signals must not only have the same sample frequency, but they must also be in sync. All devices connected to the Hammerfall 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!

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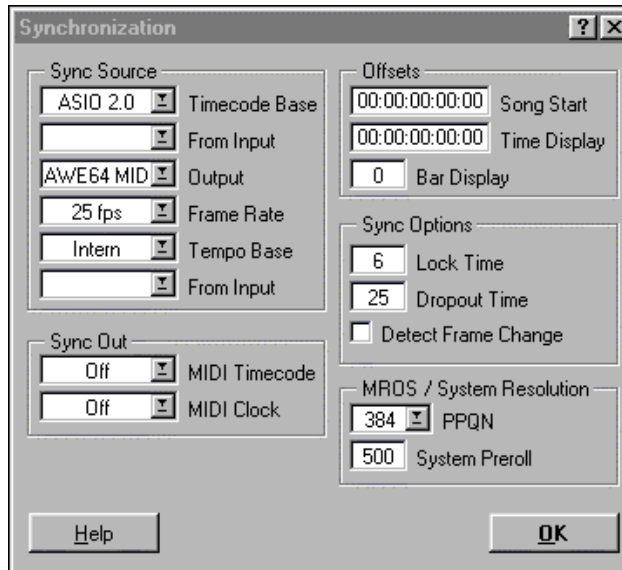
If synchronization is not working i.e. Cubase does not respond when the ADAT is set to 'Play', please try the following:

- 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*

14. Operation under GSIF (Gigasampler Interface)

14.1 Windows 98/SE/ME

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

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

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

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

14.2 Windows 2000/XP

The GSIF interface of the Hammerfall DSP allows direct operation with Gigastudio, with up to 18 channels, 96 kHz and 24 bit.

Gigastudio requires a lot of the computer's calculation power. An optimum performance is achieved with a stand-alone GSIF PC. However, when using the Hammerfall DSP, the latency is always the same as the one selected for ASIO operation. This can cause performance problems on slower machines when using GSIF and ASIO at the same time.

Please note that the W2k/XP driver fully supports multi-client operation, including the combination MME/ASIO. So for example Cubase, Gigastudio and Sonar can be used simultaneously, provided each of these programs uses its own audio channels exclusively. For example ASIO could use channels 1/2 and Gigastudio (with GSIF) channels 3/4 simultaneously, and so on.

Simultaneous operation of GSIF and ASIO requires to use different channels. For example, if Cubase uses tracks 1/2 these tracks can not be used by Gigastudio.

Common Problems

Please note that Gigastudio is running unexpectedly in the background (thus blocking its assigned audio channels), as soon as the Gigastudio MIDI ports are used – even when Gigastudio itself hasn't been started. This causes a lot of confusion, as the driver seems to behave completely buggy, and the user does not recognize the simple reason for it – for example simultaneous operation of ASIO and GSIF on the same channels.

In case Gigastudio loads well, load gig files too, but won't play at all even when using the virtual keyboard: Go to Hardware/Routing and select a valid MIDI input port. Note that blank is not valid, but <none> is.

15. 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. Some typical applications for TotalMix are:

* 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)

The block diagram of the TotalMix mixer of the HDSP 9652 shows that the record signal always stays un-altered, 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.

RME TotalMix
Hammerfall DSP Mixer
Block Diagram, 07/2001
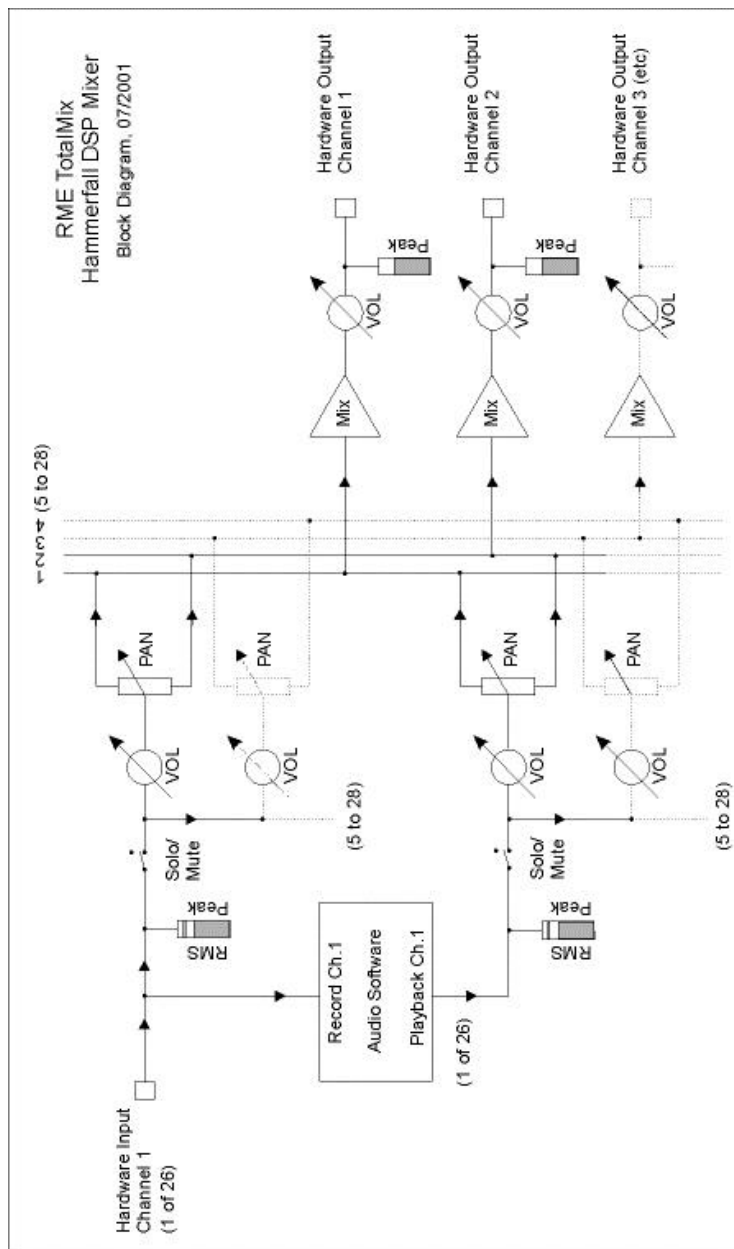
15.1 Elements of the Surface

The visible design of the TotalMix 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. Selecting one or more channels is done by clicking on the white label which turns yellow then.

15.2 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.
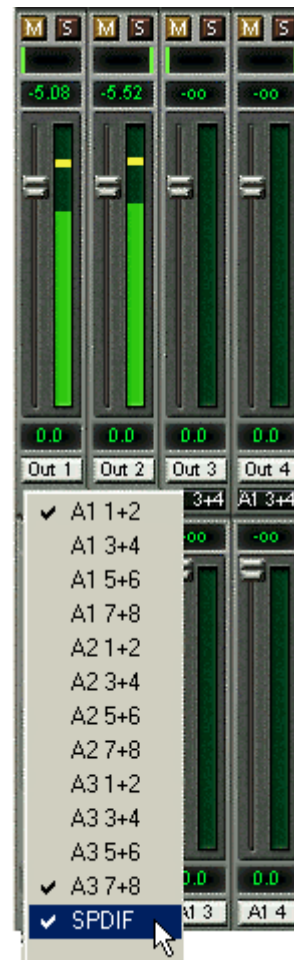
15.3 Submix View

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.

15.4 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.

15.5 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'.
- Clicking a Preset button while holding down Ctrl, the original (factory) preset will be loaded.
- Using Ctrl and any number between 1 and 8 (not on the numeric keypad!) will load the corresponding factory default preset
- Clicking the Card 2 button while holding down Ctrl opens a second TotalMix window.

The faders can also be moved pairwise, corresponding to the stereo-routing settings. This can be achieved by pressing the Alt-key and is especially comfortable when setting the SPDIF and analog output level. Even the Panoramas can be operated with Alt, from stereo through mono to inversed channels. But also the Mute and Solo buttons (ganged or inversed switching!).

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 via F2 (as in DIGICheck and the Meter Bridge).

Hotkey M toggles Master Mute on/off (and with this performs a global mute on/off). Hotkey X toggles the Matrix view on/off (see chapter 16), hotkey T the mixer view.

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

15.6 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 vertically. This way TotalMix saves space on the desktop/screen, if you 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 (Multiface, Digiface, HDSP9652 or 9632). Holding down Ctrl while clicking on button Card2 will open a second window, instead of replacing the current window content.

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, and all fader settings remain stored (even the invisible ones).

15.7 Presets

TotalMix includes 8 factory presets, stored within the program. But the presets can be changed at any time, because TotalMix stores and reads the changed presets from the files **preset11.mix** to **preset81.mix**. These files are found in the hidden directory *Documents and Settings, <Username>, Local Settings, Application Data, RME TotalMix.* The first number indicates the current preset, the second number the current card/system.

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.

The original factory preset can be reloaded by holding down the Ctrl-key and clicking on any preset button. Alternatively the files described above can be renamed, moved to a different directory, or being deleted.

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.

Note: This preset is default for ZLM and MME Mix/Replace monitoring. The factory preset 4 will also be loaded by a click on *Load Def*.

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)

15.8 Monitor

The Monitor section of the Quick Access Panel is only valid for our Windows MME driver, i.e. when using programs like WaveLab, Soundforge, Sonar or Samplitude.

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

☞ *The basic method used is as simple as it is clever: ZLM and record are controlling the Mute buttons of the corresponding channels. For this to work, Mute must be activated on the record's Input channel. The fader must not be set to m.a..*

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

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

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

ZLM is a special function for tape machine style monitoring when doing punch-ins and outs. For this to work the option 'Hardware monitoring during punch' has to be activated in Samplitude/Sequoia. Then at each punch-in the corresponding Mute buttons will be deactivated, at punch-out they will be re-activated.

All settings can be changed and configured in real-time.
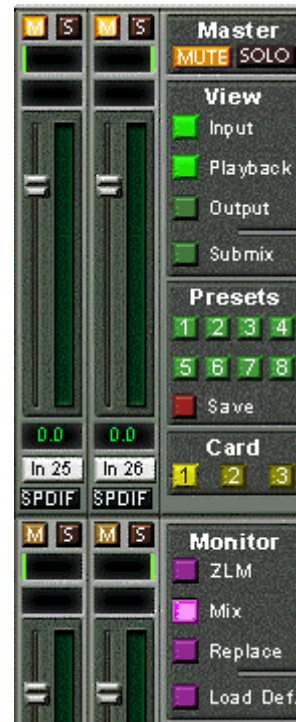
15.9 Menu Options

Always on Top: When active (checked) the window of TotalMix will always be on top of the Windows desktop.
Note: This function may result in problems with windows containing help text, as the TotalMix window will even be on top of those windows, so the help text isn't readable.

Deactivate Screensaver: When active (checked) any activated Windows screensaver will be disabled temporarily.

Ignore Position: When active, the windows size and position stored in a file or prest will not be used. The routing will be activated, but the window will not change

ASIO Direct Monitoring: When de-activated any ADM commands will be ignored by TotalMix. In other words, ASIO Direct Monitoring is globally de-activated.

15.10 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* (chapter 19.2). Nevertheless they already include many useful functions.

Peak and RMS is displayed for every channel. 'Level Meter Setup' (Menu Options or F2) 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 HDSP 9652 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'.*

16. The Matrix

The mixer window of TotalMix looks and operates similar to mixing desks, as it is based on a conventional stereo design. The matrix display presents a different method of assigning and routing channels, based on a single channel or monaural design. The matrix view of the HDSP mixer looks and works like a conventional patchbay, adding functionality way beyond comparable hardware and software soutions. While most patchbays will allow you to connect inputs to outputs with just the original level (1:1, or 0 dB, as known from mechanical patchbays), TotalMix allows you to use a freely definable gain value per crosspoint.

Matrix and TotalMix are different ways of displaying the same processes. Because of this both views are always fully synchronized. Each change in one view is immediately reflected in the other view as well.

16.1 Elements of the Surface

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

- Horizontal labels: All hardware outputs

- Vertical labels: All hardware inputs. Below are all playback channels (software playback)

- Green 0.0 dB field. Standard 1:1 routing

- Black gain field. Shows the current gain value as dB

- Orange gain field. This routing is muted.

To maintain overview when the window size has been reduced, the left and upper labels are floating. They won't left the visible area when scrolling.

16.2 Operation

Using the Matrix is a breeze. It is very easy to indentify the current crosspoint, because the labels light up in orange according to the mouse position.

If input 1 is to be routed to output 1, use the mouse and click one time on crosspoint **In1** / **1**. The green 0.0 dB field pops in, another click removes it. To change the gain (equals the use of a different fader position, see simultaneous display of the mixer view), hold Ctrl down and drag the mouse up or down, starting from the gain field. The value within the field changes accordingly. The corresponding fader in the mixer view is moving simultaneously, in case the currently modified routing is visible.

Note the difference between the left side, representing the inputs and software playback sources, and the upper side, representing the hardware outputs. If you move a fader in row 1 or 2 in TotalMix, only the specific levels (max. 2) of this routing will change in the Matrix. But moving a fader in row 3 will make all vertically activated levels move at once (for example 63/64, analog output).

A gain field marked orange indicates activated Mute status an. Mute can only be changed in the mixer view.

16.3 Advantages of the Matrix

The Matrix not always replaces the mixer view, but it significantly enhances the routing capabilities and - more important - is a brilliant way to get a fast overview on all active routings. It shows you in a glance what's going on. And since the Matrix operates monaural, it is very easy to set up specific routings with specific gains.

Example 1: You want TotalMix to route all software outputs to all corresponding hardware outputs, and have a submix of all software outputs on the analog output (equals factory preset 5). Setting up such a submix is fast and easy. But how to check at a later time, that all settings are still exactly the way you wanted them to be?

So far the only way to check that TotalMix is correctly set up this way, is to activate Submix view, step through all existing software outputs, and have a very concentrated look at the faders and displayed levels of each routing. That doesn't sound comfortably nor error-free, right? Here is where the Matrix shines. In the Matrix view, you simply see a line from upper left to lower right, all fields marked as unity gain. Plus two rows vertically all at the same level setting. You just need 2 seconds to be sure no unwanted routing is active anywhere, and that all levels match precisely!

Example 2: The Matrix allows you to set up routings which would be nearly impossible to achieve by fiddling around with level and pan. Let's say you want to send input 1 to output 1 at 0 dB, to output 2 at -3 dB, to output 3 at -6 dB and to output 4 at -9 dB. Each time you set up the right channel (2/4), the change in pan destroys the gain setting of the left channel (1/2). A real hassle! In Matrix view, you simply click on the corresponding routing point, set the level via Ctrl-mouse, and move on. You can see in the desk view how level and pan changes automatically when performing the second (fourth...) setting.

17. TotalMix Super-Features

17.1 ASIO Direct Monitoring

Start Samplitude, Sequoia, Cubase or Nuendo and TotalMix. Activate ADM (ASIO Direct Monitoring), and move a fader in the ASIO host. Now watch the corresponding fader in TotalMix magically move too. TotalMix reflects all ADM gain and pan changes in realtime. Please note that faders only move when the currently activated routing (currently visible routing) corresponds to the one in the ASIO host. Also note that the Matrix will show any change, as it shows all possible routings in one view.

With this TotalMix has become a wonderful debugging tool for ADM. Just move the host's fader and pan, and see what kind of ADM commands TotalMix receives.

The hardware output row faders are included in all gain calculation, in every possible way. Example: you have lowered the output level of a submix, or just a specific channel, by some dB. The audio signal passed through via ADM will be attenuated by the value set in the third row.

17.2 Selection and Group-based Operation

Click on the white name label of channel 1 and 2 in TotalMix. Be sure to have channel 3's fader set to a different position and click on its label too. All three labels have changed to the colour orange, which means they are selected. Now moving any of these faders will make the other faders move too. This is called 'building a group of faders', or ganging faders, maintaining their relative position.

Building groups or ganging can be done in any row, but is limited to operate horizontally within one row. If you usually don't need this, you can at least gang the analog outputs. The advantage over holding the Alt-key is that Alt sets both channels to the same level (can be handy too), while grouping via selection will retain any offset (if you need one channel to be louder all the time etc.).

Note: if you move the mouse so that any channel reaches upper or lower maximum position, and release the mouse button, the relative position is lost.

Tip: gang some submixes and watch all routing levels change like crazy in the Matrix view.

17.3 Copy Routings to other Channels

TotalMix allows to copy complete routing schemes of inputs and outputs.

Example 1: You have input 5 (guitar) routed within several submixes/hardware outputs (= headphones). Now you'll get another input with keyboards that should appear in the same way on all headphones. Select input 5, open the menu *Edit*. It shows 'Copy In 5'. Now select the desired new input, for example In 8. The menu now shows 'Paste In 5 to In 8'. Click on it - done. If you are familiar with this functionality just use Ctrl-C and Ctrl-V. Else the self updating menu will always let you know what actually will happen.

Tip: have the Matrix view open when doing this. It will show the new routings immediately, so copying is easier to understand and to follow.

Example 2: You have built a comprehensive submix on outputs 4/5, but now need the exact same signal also on the outputs 6/7. Click on Out 4, Ctrl-C, click on Out 6, Ctrl-V, same with 5/7 - you're done!

The Matrix shows you the difference between both examples. Example 1 means copying lines (horizontally), while example 2 means copying rows (vertically).

Example 3: Let's say the guitarist finished his recording, and you now need the same signal again on all headphones, but this time it comes from the recording software (playback row). No problem, you can even copy between rows 1 and 2 (copying between row 3 and 1/2 isn't possible).

But how to select while a group is active? De-selecting the group first? Not necessary! TotalMix always updates the copy and paste process with the *last* selection. This way you don't have to de-activate any group-selections when desiring to perform a copy and paste action.

18. Hotline - Troubleshooting

18.1 General

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

The ADAT timecode is not in sync

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

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

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

The input signal cannot be monitored in real-time

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

The first 8 channels don't seem to work

- SPDIF output has been switched to ADAT1. This means that the first ADAT output device, and therefore the first 8 channels in the ASIO application, are no longer available. All channels and their assignments still exist, but the optical transmitter has been disconnected from the ADAT and is now fed from the SPDIF output (channels 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.
- Activate Busmaster mode for the hard disks.
- In case of a recently done BIOS update of the motherboard: Propably 'Load BIOS Defaults' was loaded instead of 'Load Setup Defaults'. This sets the 'PCI Latency Timer' to 0 (default: 32).

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

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

18.2 Installation

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

HDSP 9652 is found in the Device Manager (>Settings/Control Panel/System<), category 'Sound-, Video- and Gamecontroller'. A double click on ' HDSP 9652' starts the properties dialog. Choosing 'Resources' shows Interrupt and Memory Range.

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

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

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

19. HDSP Software

19.1 DIGICheck 4.0

The DIGICheck software is a unique utility developed for testing, measuring and analysing digital audio streams. Although the DIGICheck software is fairly self-explanatory, it still includes a comprehensive online help. DIGICheck 4.0 operates as multi-client ASIO host, and can therefore be used in parallel to any software, be it MME, ASIO or GSIF, both inputs and even outputs (!).

The following is a short summary of the available functions:

- **Level Meter**. High precision 24-bit resolution, 2/8/26 channels. Application examples: Peak level measurement, RMS level measurement, over-detection, phase correlation measurement, dynamic range and signal-to-noise ratios, RMS to peak difference (loudness), long term peak measurement, input check. Oversampling mode for levels higher than 0 dBFS.
- **Vector Audio Scope**. World wide unique Goniometer showing the typical afterglow of a oscilloscope-tube. Includes Correlation meter and level meter.
- **Spectral Analyser.** World wide unique 10-, 20- or 30-band display in analog bandpass-filter technology. 192 kHz-capable!
- **Bit Statistics & Noise**. Shows the true resolution of audio signals as well as errors and DC offset. Includes Signal to Noise measurement in dB and dBA, plus DC measurement.
- **Totalyser**. Spectral Analyser and Vector Audio Scope as one window function.
- **Channel Status Display**. Detailed analyzis and display of SPDIF and AES/EBU Channel Status data.
- **Completely Multi-client.** Open as many measurement windows as you like, on any channels and inputs or outputs!

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

19.2 HDSP Meter Bridge

The Hammerfall DSP Meter Bridge is a unique, highly flexible and handy tool for level metering. As opposed to DIGICheck, the HDSP Meter Bridge receives all level data directly from the hardware. Thus the Meter Bridge can run in parallel to any other program. As Peak, Over and RMS calculations are performed directly in hardware, the CPU load caused is limited to the graphics routines – and is near zero on todays computers.

Although the HDSP Meter Bridge is fairly self-explanatory, it still includes a comprehensive online help. To start press F2 and F3, the most important hotkeys.

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

20. Accessories

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

Part Number	Description
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

21. 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, Cubase, Cakewalk, Sonar and SAWPlus32
Step by step instructions for use with RME cards.

DIGICheck: Analysis, tests and measurements with the DIGI96 series
A description of DIGICheck, including technical basics.

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

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

22. 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 Audio AG apply at all times.

23. Appendix

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

http://www.rme-audio.com

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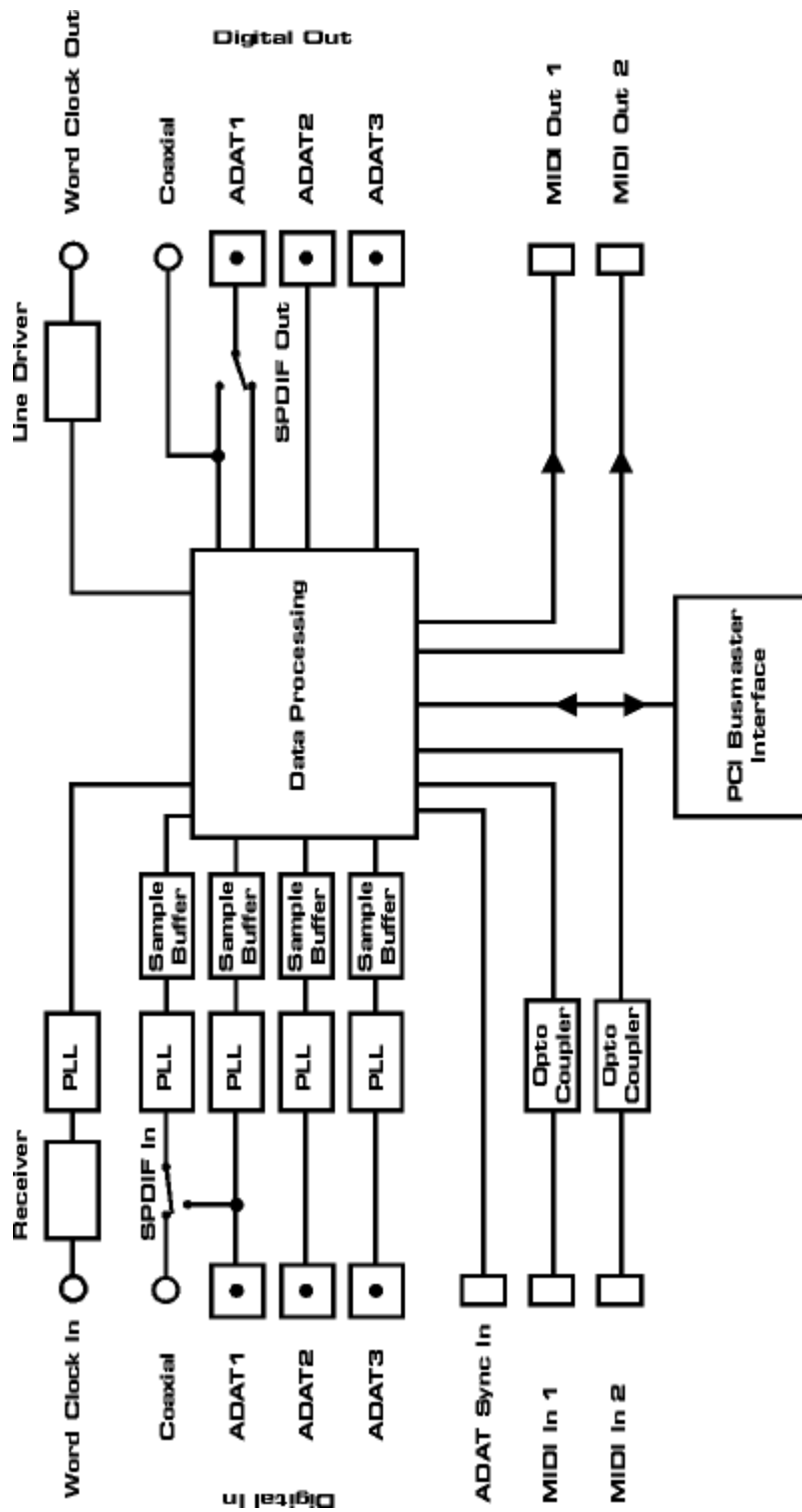

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24. Diagrams

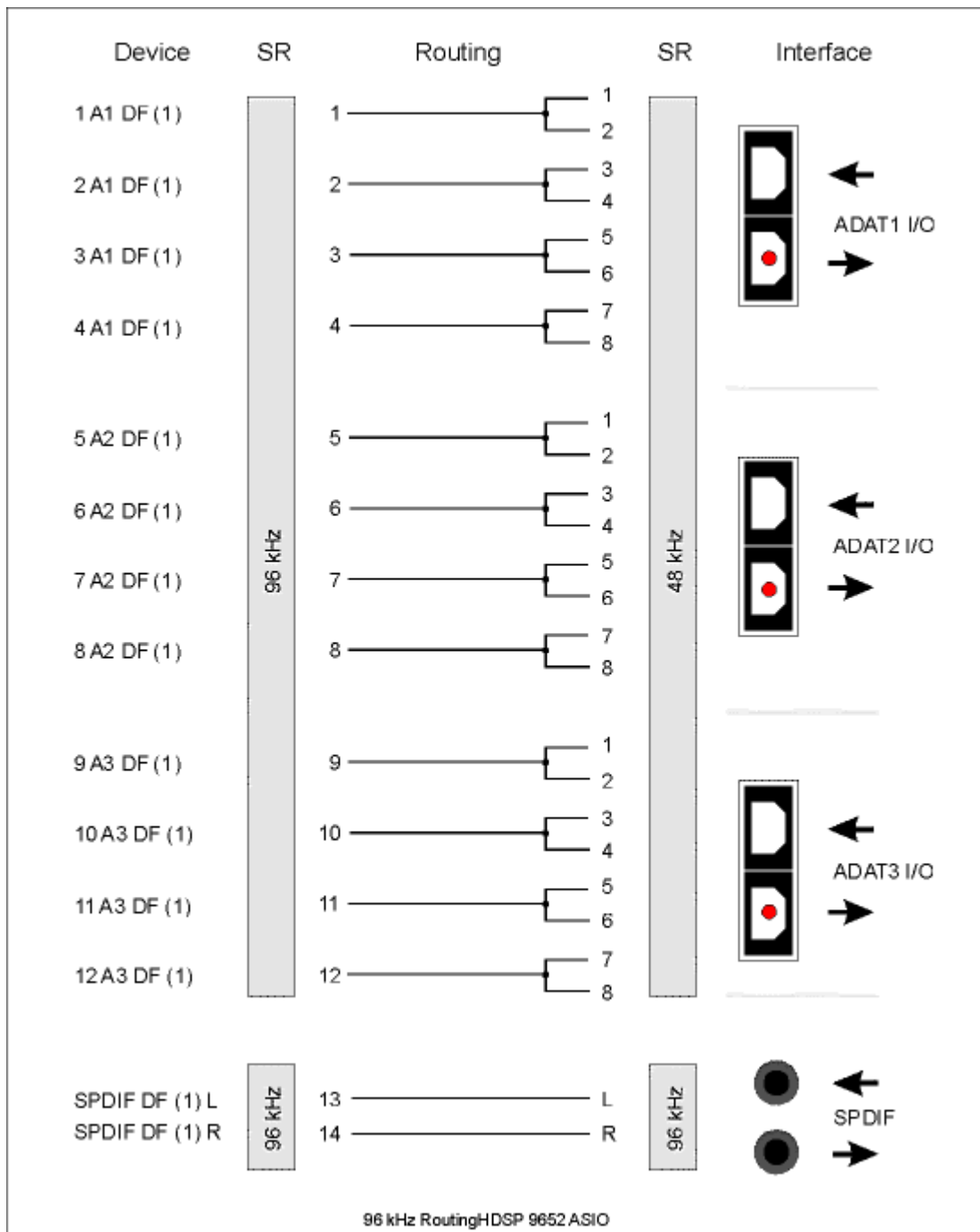## 24.1 Block Diagram HDSP 9652

24.2 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


96 kHz RoutingHDSP 9652 ASIO

24.3 ADAT Track Routing, MME at 96 kHz

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

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

96 kHz Routing HDSP 9652 MME
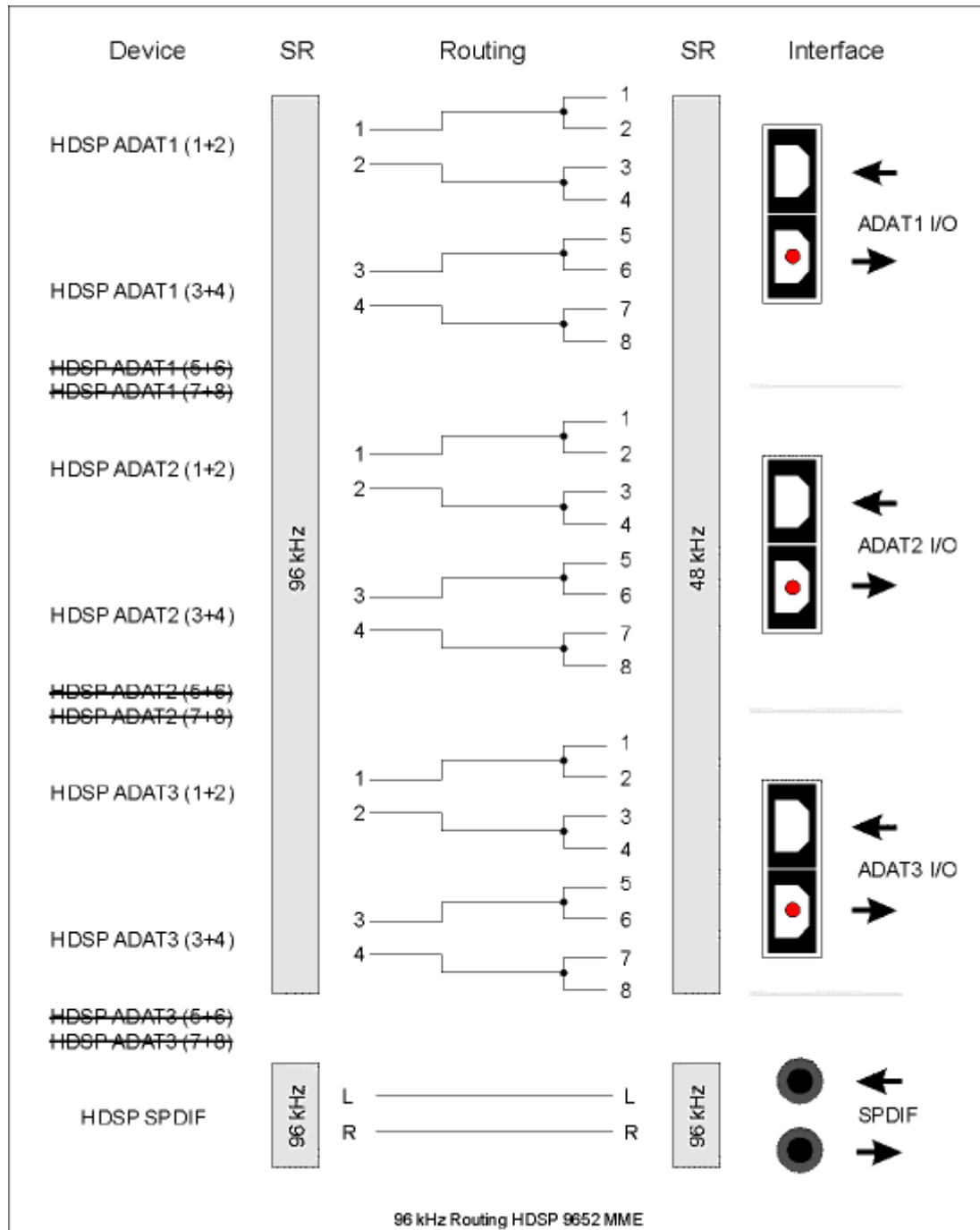
25. CE and FCC Compliance Statements

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.