User's Guide

RME
INTELLIGENT AUDIO SOLUTIONS

Hammerfall® DSP System

HDSP 9632

DVD 24 Bit / 192 kHz ready

TotalMix™

ASIO

SyncAlign®

ZLM®

SyncCheck®

SteadyClock™

adat® OPTICAL

PCI Busmaster Digital I/O Card
2 + 8 + 2 Channels SPDIF / ADAT / Analog Interface
24 Bit / 192 kHz Digital Audio
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. With *SteadyClock*, the HDSP 9632 excels in analog inputs and outputs with unsurpassed quality. The latest Plug and Play technology guarantees a simple installation, even for the inexperienced user. 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 2000/XP and Mac OS X. An ALSA driver for Linux is also available (see chapter 7.5).

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 9632 PCI card
- Quick Info guide
- RME Driver CD
- Digital adapter cable (phono / phono to D-type 9 pin)
- Analog adapter cable (phono / phono / TRS / MIDI to D-type 15 pin)
- Internal cable (2-core)
- 1 optical cable (TOSLINK)

3. System Requirements

- Windows 2000/XP, Linux, Mac OS 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 32 ASIO channels
- All settings can be changed in real-time
- Enhanced mixed mode: All inputs and all outputs can be used simultaneously
- 8 available buffer sizes/latencies: 1.5 / 3 / 6 / 12 / 23 / 46 / 93 / 186 ms
- Automatic and intelligent master/slave clock control
- 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
- 1 x MIDI I/O, 16 channels high-speed MIDI
- DIGICheck DSP: Level meter in hardware, peak- and RMS calculation
- TotalMix: 512-channel mixer with 40 bit internal resolution
- SteadyClock: jitter-immune, super stable digital clock

5. Technical Specifications

5.1 Analog

AD – Line In

- Resolution AD: 24 Bit
- Signal to Noise ratio (SNR): 109 dB RMS unweighted, 111 dBA @ 44.1 kHz
- THD @ -3 dBFS: -101 dB, < 0.001 %
- THD+N @ -3 dBFS: -99 dB, < 0.0015 %
- Crosstalk: 108 dB
- Frequency response AD @ 44.1 kHz, -0.5 dB: 5 Hz - 21.5 kHz
- Frequency response AD @ 96 kHz, -0.5 dB: 5 Hz – 45.3 kHz
- Frequency response AD @ 192 kHz, -1 dB: 5 Hz - 74 kHz
- Input Line: phono unbalanced, optional XLR balanced
- Input impedance: 10 kOhm
- Input sensitivity: Lo Gain, +4 dBu, -10 dBV
- Input level for 0 dBFS @ Lo Gain: +19 dBu
- Input level for 0 dBFS @ +4 dBu: +13 dBu
- Input level for 0 dBFS @ -10 dBV: +2 dBV

DA – Line Out

- Resolution DA: 24 Bit
- Signal to Noise ratio (SNR): 110 dB RMS unweighted, 112 dBA @ 44.1 kHz (unmuted)
- THD: < - 104 dB, < 0.00063 %
- THD+N: < -102 dB, < 0.0008 %
- Crosstalk: > 110 dB
- Maximum output level: +19 dBu
- Frequency response @ 44.1 kHz, -0.5 dB: 1 Hz – 21.1 kHz
- Frequency response @ 96 kHz, -0.5 dB: 1 Hz – 43.5 kHz
- Frequency response @ 192 kHz, -0.5 dB: 1 Hz – 70 kHz
- Output Line: phono unbalanced, optional XLR balanced
- Output impedance: 50 Ohm
- Output level: Hi Gain, +4 dBu, -10 dBV
- Output level at 0 dBFS @ Hi Gain: +19 dBu
- Output level at 0 dBFS @ +4 dBu: +13 dBu
- Output level at 0 dBFS @ -10 dBV: +2 dBV

DA - Stereo Monitor Output (Phones)

- Signal to Noise ratio (SNR): 110 dB RMS unweighted, 112 dBA @ 44.1 kHz (unmuted)
- THD: < - 100 dB, < 0.001 %
- THD+N: < -98 dB, < 0.0015 %
- Crosstalk: > 100 dB
- Frequency response @ 44.1 kHz, -0.5 dB: 1 Hz – 21.1 kHz
- Frequency response @ 96 kHz, -0.5 dB: 1 Hz – 43.5 kHz
- Frequency response @ 192 kHz, -0.5 dB: 1 Hz – 70 kHz
- Output: 6.3 mm TRS jack, optional Neutrik lockable
- Output impedance: 50 Ohm
- Output level @ 0 dB: +13 dBu
- Output level @ –6 dB: +7 dBu
- Output level @ -12 dB: +1 dBu

5.2 Digital

- Clocks: Internal, ADAT In, SPDIF In, optional Word Clock In
- Low jitter design: < 1 ns in PLL mode, all inputs
- Internal clock: 800 ps jitter, random spread spectrum
- Jitter suppression of external clocks: about 30 dB (2.4 kHz)
- Effective clock jitter influence on AD and DA conversion: near zero
- PLL ensures zero dropout, even at more than 100 ns jitter
- Additional Bitclock PLL for trouble-free varispeed ADAT operation
- Sample frequencies: 32 / 44.1 / 48 / 64 / 88.2 / 96 / 128 / 176,4 / 192 kHz

5.3 Digital Interface

- Digital inputs and outputs ground-free transformer coupled
- Standard: optical (TOSLINK), phono (SPDIF), optional XLR (AES/EBU)
- High-sensitivity input stage phono/XLR: < 0.2 Vss input level
- Output voltage phono: 0.8 V, XLR: 3.5 V

5.4 MIDI

- 1 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

5.5 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)

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 9632 from its protective bag, discharge any static in your body by touching the metal chassis of the PC.
4. Prior to installation: Connect the HDSP 9632 card to any Expansion Board (if present) using the supplied flat ribbon cable. Please read the Expansion Board's manual for more details.
5. Insert the HDSP 9632 firmly into a free PCI slot, press and fasten the screw.
6. Replace the computer's housing.
7. Reconnect all cables including the power cord.

7. Driver Installation

7.1 Windows 2000/XP

After the PCI card has been installed correctly 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 9632 is ready for use.

HDSP 9632 can be easily configured using the HDSP system’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!

7.2 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 HDSP 9632/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.

7.3 Flash Update

The Flash Update Tool updates the HDSP 9632'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 9632 (150 or up), 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 is 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 7.2.

7.4 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'.

7.5 Linux/Unix

An ALSA driver for Linux is available soon. TotalMix has been ported to Linux too. Further information on ALSA is available at

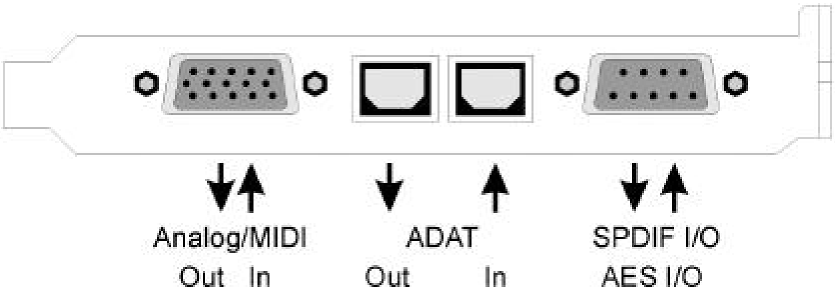
http://www.alsa-project.org

8. Operation and Usage

8.1 External Connections

HDSP 9632 consists of the main PCI board and optional Expansion Boards.

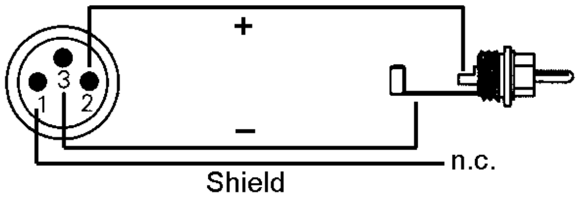
The main board's bracket has one ADAT optical input, a 9-pin D-type socket and a 15-pin D-type socket. Coaxial SPDIF input and output is provided via the included adapter cable, whereby the red phono socket is the output.



The ADAT I/O can also be used as SPDIF optical I/O, when this mode is selected in the Settings dialog. The Settings dialog is started by clicking on the hammer symbol in the system tray. The card 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 to connect two devices, i.e. using the HDSP 9632 as a splitter.

In case the optional digital XLR breakout cable is not used, receiving signals in AES/EBU format require a cable adapter. 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.

HDSP 9632 offers balanced line inputs when the optional analog XLR breakout cable is used. The electronic input stage is built in a servo balanced design which handles balanced and unbalanced connections correctly.

When using unbalanced cables with stereo TRS jacks, the 'ring' contact of the cable's jack should be connected to pin 1 (ground). Otherwise noise may occur, caused by the unconnected negative input of the balanced input.

Optional HDSP 9632 Word Clock Module
The Expansion Board *HDSP 9632 Word Clock Module* provides one word clock input and two word clock outputs. A green LED signals the *LOCK* state of the word clock input stage. A small push switch allows to activate 75 Ohm termination for the word clock input. The green LED is lit when termination is active.

8.2 Internal Connections

CD / AEB / SYNC 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.

ADAT 1 OUT
This internal ADAT output carries the same audio data as the optical output 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.

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.

WORD CLOCK MODULE
10-pin connector for the optional *HDSP 9632 Word Clock Module* via flat ribbon cable.

AI4S-192 / AO4S-192
26-pin connector for the optional analog expansion boards, AI4S-192 and AO4S-192.

Blue Jumper
This is no internal input or output. The blue jumper allows to change the power-on delay when booting the computer. Factory default is jumper in place. The FPGA of the HDSP 9632 will then be booted about 0.5 seconds after switching on the computer. This provides an effective way to get rid of problems caused by unstable power supplies in the moment of power-on. In case this delay will become a problem some day (very unlikely, and no system known so far), the jumper can be pulled. The delay is then reduced to 30 ms.

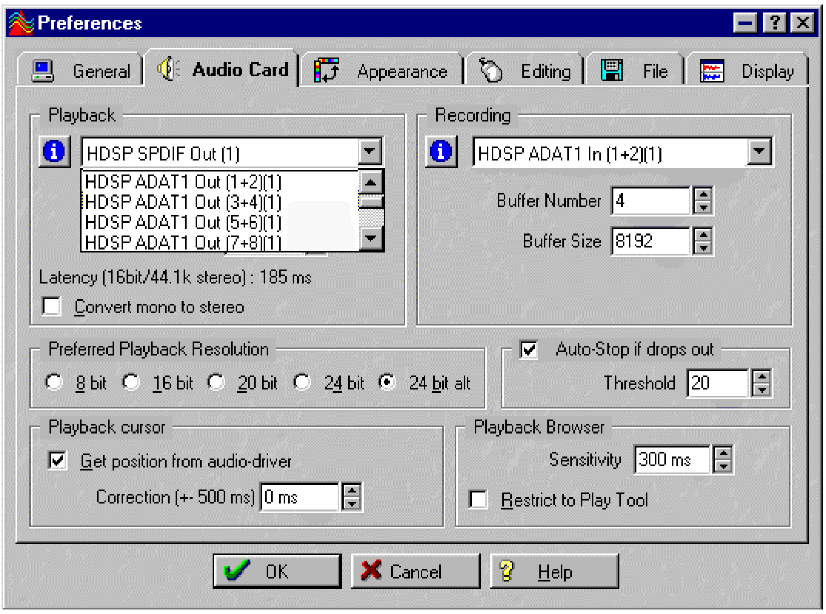
8.3 Playback

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 8 to 4. Under Windows MME, channels are routed to ADAT devices in double-speed mode as follows:

- Only stereo pairs (1+2) and (3+4) of the 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 'Channel Routing MME 96 kHz', section 25. 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 9632 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 the analog outputs of the HDSP 9632. Supported are all modes, from 2 to 8 channels, at 16 bit resolution and 48 kHz sample rate. Using the optional AO4S-192, a high quality 5.1 playback is possible, as 6 channels are available then. Playback via the ADAT output of the HDSP 9632 is also supported.

For this to work an output wave device of the HDSP 9632 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 ADAT 3/4 and 6-channel mode, playback will happen on channels 3 to 8. Choosing ADAT 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 Analog 1/2, the 6-channel mode will only be available when an AO4S-192 is installed.

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 with Windows 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 (or trash the motherboard…).

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 Digital Recording

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

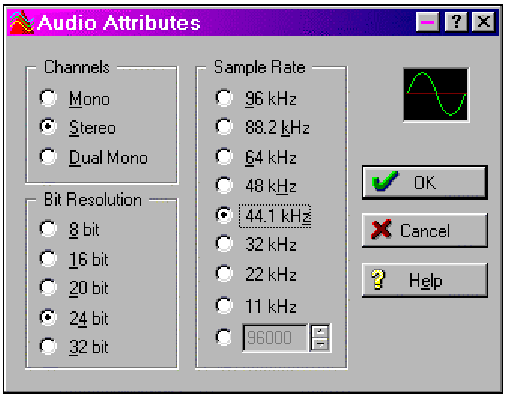
To take this into account, RME have added two unique features to the HDSP 9632: a comprehensive I/O signal status display (showing sample frequency, lock and sync status) in the Settings dialog, and the protective *Check Input* function.

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 – letting the system behave like a tape machine. This method has been implemented in all versions of Samplitude and Sequoia (by Magix), and can be activated using the global track option 'Hardware monitoring during Punch'.

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

8.8 Recording analog

For recordings via the analog inputs the corresponding record device has to be chosen (HDSP Analog (x+x)). The desired sample rate has to be set within the recording software.

The input sensitivity of the analog input(s) can be changed via the Settings dialog, to meet the most often used studio levels, see next section.

8.9 Analog Inputs

The HDSP 9632 provides unbalanced phono line inputs. The optional analog XLR breakout cable turns the inputs into fully balanced ones via its XLR connectors.

The electronic input stage is built in a servo balanced design which handles unbalanced (mono jacks) and balanced (stereo jacks) correctly, automatically adjusting the level reference.

When using unbalanced cables with the XLR breakout cable: be sure to connect the 'ring' contact of a stereo TRS jack, and pin 3 of a XLR jack, to ground. Otherwise noise may occur, caused by the unconnected negative input of the balanced input.

One of the main issues when working with an AD-converter is to maintain the full dynamic range within the best operating level. Therefore the HDSP 9632 internally uses hi-quality electronic switches, which allow for a perfect adaptation of all inputs to the three most often used studio levels.

The 'standardized' studio levels do not result in a (often desired) full scale level, but take some additional digital headroom into consideration. The amount of headroom is different in different standards, and again differently implemented by different manufacturers. Because of this we decided to define the levels of the HDSP 9632 in a most compatible way.

Reference	0 dBFS @	Headroom
Lo Gain	+19 dBu	15 dB
+4 dBu	+13 dBu	9 dB
-10 dBV	+2 dBV	12 dB

With +4 dBu selected, the according headroom meets the latest EBU recommendations for Broadcast usage. At -10 dBV 12 to 15 dB headroom are common practice, each mixing desk operating at -10 dBV is able to send and receive much higher levels. Lo Gain allows to work with high levels, best suited for professional users who prefer to work balanced and at highest levels.

The above levels are also found in our ADI-8 series of AD/DA converters, the Multiface, and even in our Mic-Preamps QuadMic and OctaMic. Therefore all RME devices are fully compatible to each other.

8.10 Analog Outputs

The short circuit protected, low impedance line outputs are available as unbalanced outputs via a phono breakout cable. The optional analog XLR breakout cable provides XLR connectors and fully balanced operation.

☝ *The electronic output stage does not operate servo balanced! When connecting unbalanced equipment, make sure pin 3 of the XLR output is not connected.A conection to ground will cause a decreased THD (higher distortion)!*

☝ *When using the analog XLR breakout cable, make sure 'Breakout Cable / XLR' is selected in the Settings dialog. Else the analog output level will be 6 dB too high!*

To maintain an optimum level for devices connected to the analog outputs, the HDSP 9632 internally uses hi-quality electronic switches, which allow for a perfect adaptation of all outputs to the three most often used studio levels.

As with the analog inputs the analog output levels are defined to maintain a problem-free operation with most other devices. The headroom of the HDSP 9632 lies between 9 and 15 dB, according to the chosen reference level:

Reference	**0 dBFS @**	**Headroom**
Hi Gain	+19 dBu	15 dB
+4 dBu	+13 dBu	9 dB
-10 dBV	+2 dBV	12 dB

With +4 dBu selected, the according headroom meets the latest EBU recommendations for Broadcast usage. At -10 dBV 12 to 15 dB headroom are common practice, each mixing desk operating at -10 dBV is able to send and receive much higher levels. Lo Gain allows to work with high levels, best suited for professional users who prefer to work balanced and at highest levels.

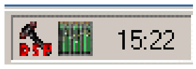
The above levels are also found in our ADI-8 series of AD/DA converters, the Multiface, and even in our Mic-Preamps QuadMic and OctaMic. Therefore all RME devices are fully compatible to each other.

9. Configuring the HDSP 9632

9.1 General Information

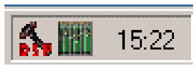
Configuring the HDSP 9632 is done via its own settings dialog. The panel 'Settings' can be opened:

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



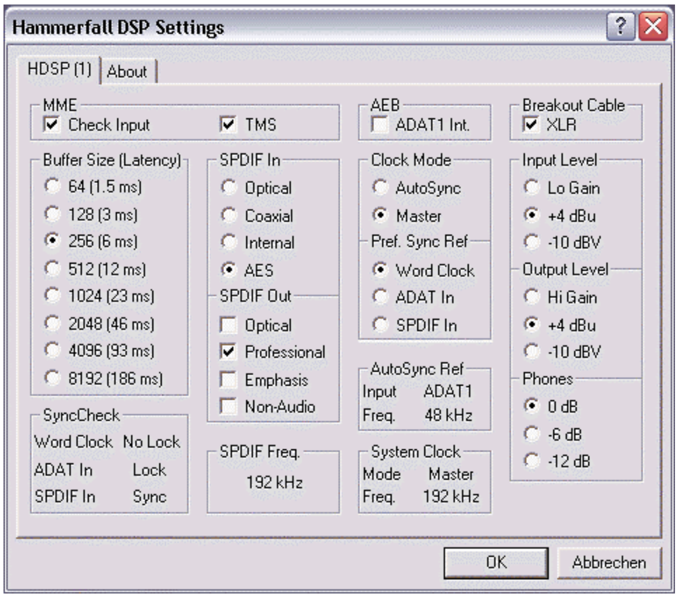
The mixer of the Hammerfall DSP System (TotalMix) can be opened:

- 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



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.

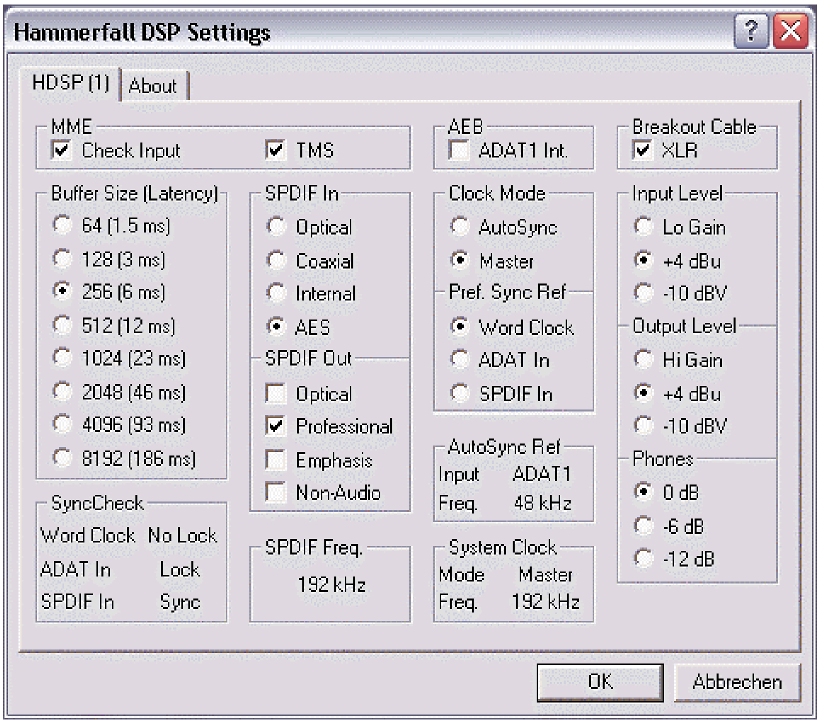
The status displays at the bottom of the dialog box give the user precise information about the current status of the system, and the status of all signals. 'SyncCheck' indicates whether there is a valid signal for each input ('Lock' or 'No Lock'), or if there is a valid *and* synchronous signal ('Sync').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

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

TMS activates the transmission of Channel Status data and Track Marker information of the SPDIF input. *Check Input* is valid for MME only.

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, 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, 'Optical' to the optical TOSLINK input, Internal to the jumper CD/SYNC IN/AEB IN, AES to the optional XLR cable.

SPDIF Out
The SPDIF output signal is constantly available at the phono plug. After selecting 'Optical' it is also routed to the optical TOSLINK output. 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 and word clock.

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.

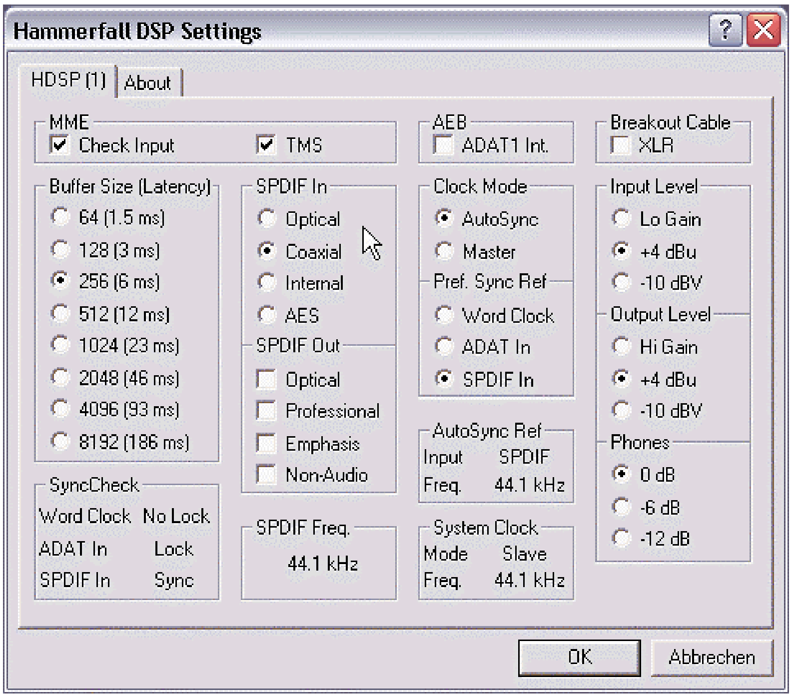
AutoSync guarantees that normal record and record-while-play will always work correctly. In certain cases however, e.g. when the inputs and outputs of a DAT machine are connected directly to the Hammerfall DSP, AutoSync may cause feedback in the digital carrier, so synchronization breaks down. To remedy 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'.

The HDSP 9632's ADAT optical input and the SPDIF input operate 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).

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 ADAT input (ADAT 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 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.

Due to the HDSP 9632's outstanding clock control a synchronization of the output signal to the input signal is not only possible at identical sample rates, but also at half, quarter, double and quad sample rates. A playback of 192 kHz can be easily synchronized via a 48 kHz clock source.

Thanks to its AutoSync technique and lightning fast PLLs, the Hammerfall DSP is not only capable of handling standard frequencies, but also any sample rate between 25 and 200 kHz. Even the word clock input, which most users will use in varispeed operation, allows any frequency between 25 kHz and 200 kHz.

At 88.2 or 96 kHz: If the ADAT input 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. Digital signals can only be processed or transferred between devices if these share the same clock. Otherwise the signals are misinterpreted, causing distortion, clicks/crackle or 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.

But word clock also has some disadvantages. The word clock is based on a fraction of the actually needed clock. For example SPDIF: 44.1 kHz word clock (a simple square wave signal) has to be multiplied by 256 inside the device using a special PLL (to about 11.2 MHz). This signal then replaces the one from the quartz crystal. Big disadvantage: because of the high multiplication factor the reconstructed clock will have great deviations called jitter. The jitter of a word clock is typically 15 times higher as when using a quartz based clock.

The end of these problems should have been the so called Superclock, which uses 256 times the word clock frequency. This equals the internal quartz frequency, so no PLL for multiplying is needed and the clock can be used directly. But reality was different, the Superclock proved to be much more critical than word clock. A square wave signal of 11 MHz distributed to several devices - this simply means to fight with high frequency technology. Reflections, cable quality, capacitive loads - at 44.1 kHz these factors may be ignored, at 11 MHz they are the end of the clock network. Additionally it was found that a PLL not only generates jitter, but also also rejects disturbances. The slow PLL works like a filter for induced and modulated frequencies above several kHz. As the Superclock is used without any filtering such a kind of jitter and noise suppression is missing. No wonder Superclock did not become a commonly accepted standard.

The actual end of these problems is offered by the **SteadyClock** technology of the HDSP 9632. Combining the advantages of modern and fastest digital technology with analog filter techniques, re-gaining a low jitter clock signal of 11 MHz from a slow word clock of 44.1 kHz is no problem anymore. Additionally, jitter on the input signal is highly rejected, so that even in real world usage the re-gained clock signal is of highest quality.

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 examples where 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 – it will also work as long as the termination resistors are 75 Ohm.

The HDSP 9632's word clock input on the optional *HDSP 9632 Word Clock Module* can be high-impedance or terminated, ensuring maximum flexibility. If termination is necessary (e.g. because HDSP 9632 is the last device in the chain), push the WCM's termination switch so that the yellow termination LED lights up.

In case the HDSP 9632 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. The last device in the chain should be terminated using another T-adapter and a 75 Ohm resistor (available as short BNC plug). Devices with internal termination do not need T-adaptor and terminator plug.

Due to the outstanding SteadyClock technology of the HDSP 9632, we recommend not to pass the input signal via T-adapter, but to use the WCM's two word clock outputs. Thanks to SteadyClock, the input signal will both be freed from jitter and - in case of loss or drop out – be held at the last valid frequency.

10.3 General Operation

The green 'Lock' LED at the Expansion Board will light up when the input detects 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.

Like all ADAT ports, the word clock output operates in Single Speed mode only. 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/HDSP 9632). 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 dialog ('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, 96 kHz, 176,4 kHz, 192 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

For AES/EBU output operation the optional digital XLR breakout cable is required, as the phono output has too low voltage for a reliable operation in AES format.

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

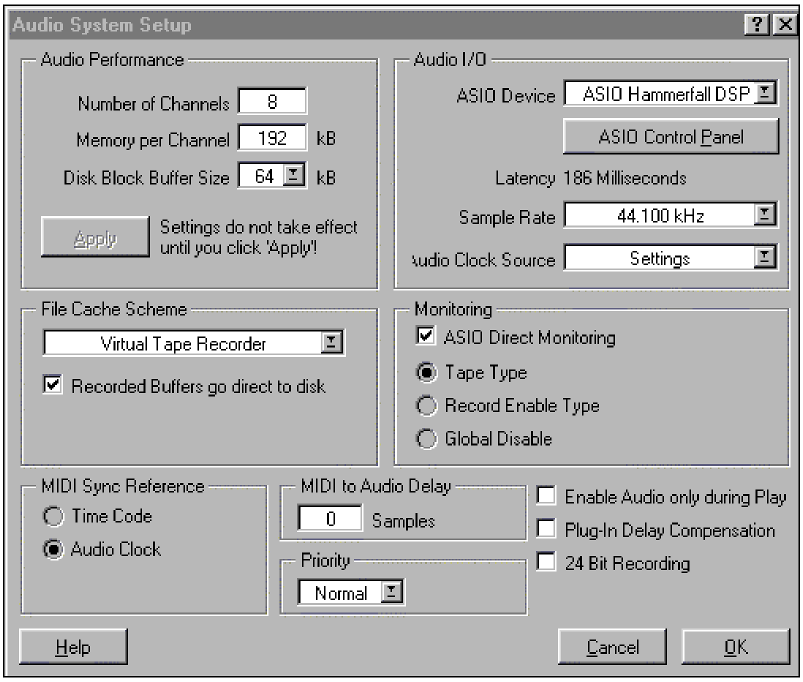
The audio bit in the header can be set to 'Non-Audio'. This is necessary when Dolby AC-3 encoded data is sent to external decoders (surround-sound receivers, television sets etc. with AC-3 digital inputs), as these decoders would otherwise not recognize the data as AC-3.

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 adaptated 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, the ADAT optical input and output operates in Sample Split mode, so the number of available channels is reduced from 8 to 4.

At a sample rate of 176.4 or 192 kHz (Quad Speed Mode) the ADAT I/O is no longer available. Nevertheless it will send out a synchronized ADAT signal at a quarter of the sample rate.

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!

14. Operation under GSIF (Gigasampler Interface)

Windows 2000/XP

The GSIF interface of the Hammerfall DSP allows direct operation with Gigastudio, with up to 16 channels, 96kHz and 24bit.

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.

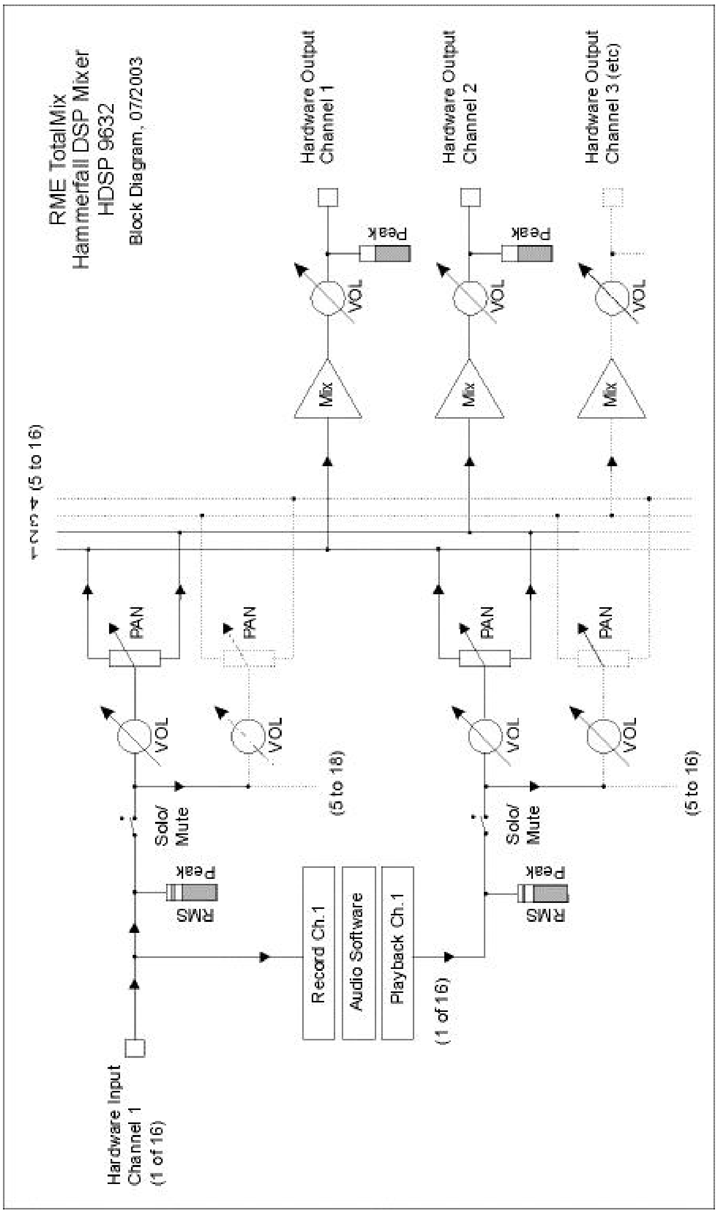
15. TotalMix: Routing and Monitoring

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.

Here are some typical applications for TotalMix:

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

The block diagram of the TotalMix mixer of the HDSP 9632 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.

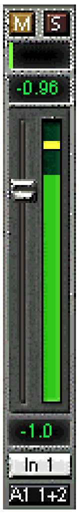


15.1 Elements of the Surface

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 large time constant, so that it shows the average loudness quite well.

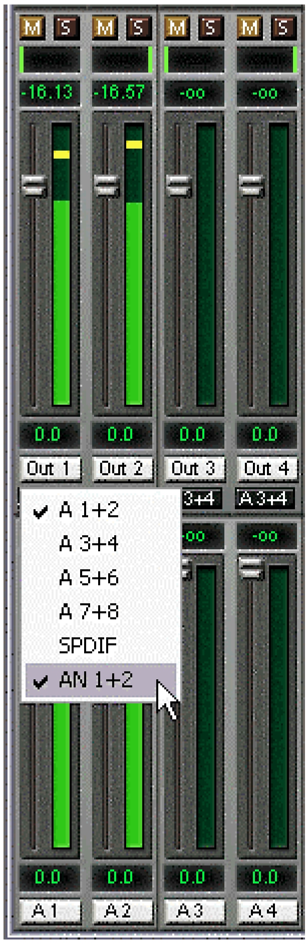
Below the fader, the current gain and panorama values are shown.

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 mixer 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 analog output. Please start a multitrack playback and connect a headphone to the phones 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' (ADAT channel 1+2). Click onto 'AN 1+2'. The list disappears, the routing window no longer shows 'A1 1+2', but 'AN 1+2'. 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 11/12 (AN 1 and 2), 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 of Out 1, in this case the routing between channels 11 and 12. 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 'A 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. Press and hold Ctrl, then click on any place of the fader area. The fader jumps to 0 dB. In the hardware output row, the fader will jump to –6 dB.

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 'A 3+4', the other one at 'AN 1+2'. Click onto 'SPDIF'. The window disappears, fader and panpot jump to their initial values, the signal can now be routed to the SPDIF output. You can continue, until all entries have got a checkmark, i. e. you can send the signal to all outputs simultaneously.

You will certainly have noticed that the mix has not changed, while you were routing the channel to other outputs and setting different gain values. With all 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 'AN 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 (up to 8 completely independent stereo submixes, 4 4-channel submixes etc.). 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. 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.

Further hotkeys are available to control the configuration of the Level Meter (see chapter 15.10):

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, HDSP 9652 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
All channels routed 1:1.

Details: All inputs maximum attenuation (m.a.). All playback channels 0 dB, routet to the same output. All outputs 0 dB. All channels prepared for all routings to left/right panning. Level display RMS +3 dB not activated.

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

Preset2.mix
Details: Submix to Analog –6 dB. As Preset 1, plus submix of all inputs and playbacks to AN 1+2. View Submix AN 1+2 active.

Preset3.mix
Details: All channels 1:1. As Preset 1, but all input faders set to 0 dB (1:1 pass through).

Preset4.mix
Details: All channels 1:1. 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
Details: All faders m.a. As Preset 1, but all outputs m.a.

Preset6.mix
Details: Submix to SPDIF at -6 dB. As Preset 1, plus submix of all inputs and playbacks to SPDIF. View Submix SPDIF active.

Preset7.mix
Details: All channels routed 1:1. As Preset 1, but Submix View AN 1+2 active.

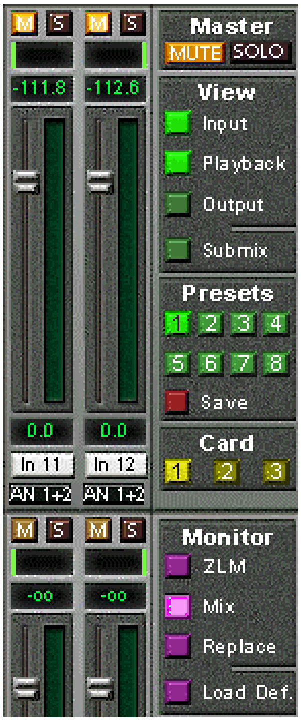
Preset8.mix
Description: Panic. As Preset 4, but also playback muted (no output signal)

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 (factory preset 4), 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.

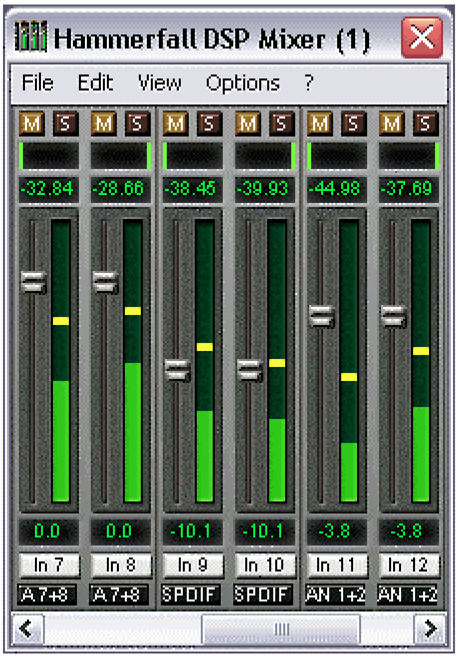
15.10 Level Meter

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 20.2). 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 9632 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'.

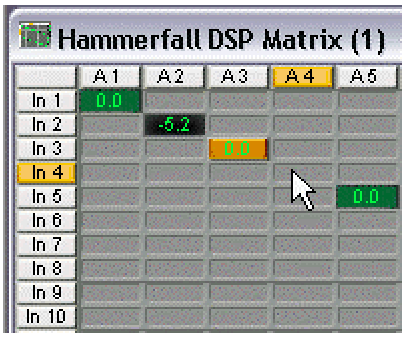
16. The Matrix

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 a 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. 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 inputs and software outputs on the analog output (equals factory preset 2). Setting up such a submix is 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).

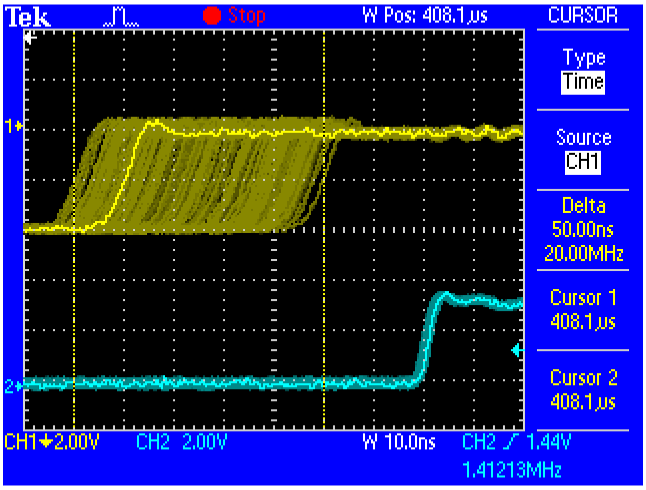
But how to select while a group is active? De-selecting the group first? Not necessary! TotalMix always updates the copy and paste process with the *last* selection. This way you don't have to de-activate any group-selections when desiring to perform a copy and paste action.

18. SteadyClock

The Hammerfall DSP 9632's SteadyClock technology guarantees an excellent performance in all clock modes. Its highly efficient jitter suppression refreshes and cleans up any clock signal, and provides it as reference clock at its digital outputs.

SteadyClock has been originally developed to gain a stable and clean clock from the heavily jittery MADI data signal (the embedded MADI clock suffers from about 80 ns jitter). Using word clock, SPDIF and ADAT, you'll most probabaly never experience such high jitter values. But SteadyClock is not only ready for them, it would handle them just on the fly.

Common jitter values in real world applications are below 10 ns, while a very good clock will have less than 2 ns.



The screnshot shows an extremely jittery SPDIF signal of about 50 ns jitter (top graph, yellow). Thanks to SteadyClock this signal turns into a clock with less than 2 ns jitter (lower graph, blue). The cleaned and jitter-freed signal can be used as reference clock for any application, without any problem. The signal processed by SteadyClock is of course not only used internally, but also available at the card's optional two word clock outputs. It is also used to clock the digital outputs.

19. Hotline - Troubleshooting

19.1 General

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

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 ADAT. 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 9/10).

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.

19.2 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 9632 is normally in the Device Manager (>Settings/Control Panel/System<), category 'Sound-, Video- and Gamecontroller'. A double click on ' HDSP 9632' 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.de, 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.

20. HDSP Software

20.1 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**. Detailled 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.

20.2 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.

21. Accessories

RME offers several optional components. Additionally parts of the HDSP 9632, like the special breakout cables, are available separately.

Part Number	**Description**
OK05	Optical cable, Toslink, 0.5 m (1.5 ft)
OK1	Optical cable, Toslink, 1 m (3.3 ft)

Standard lightpipe with TOSLINK connectors, RME approved quality.

BO9632	Breakout cable SPDIF (RCA)
BO 968	Breakout cable AES (XLR)
BO9632-CMKH	Breakout cable Analog (RCA)
BO9632-XLRMKH	Breakout cable Analog (XLR)

22. 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.

SteadyClock
Technology, background, measurements

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

23. 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.

24. Appendix

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

http://www.rme-audio.de

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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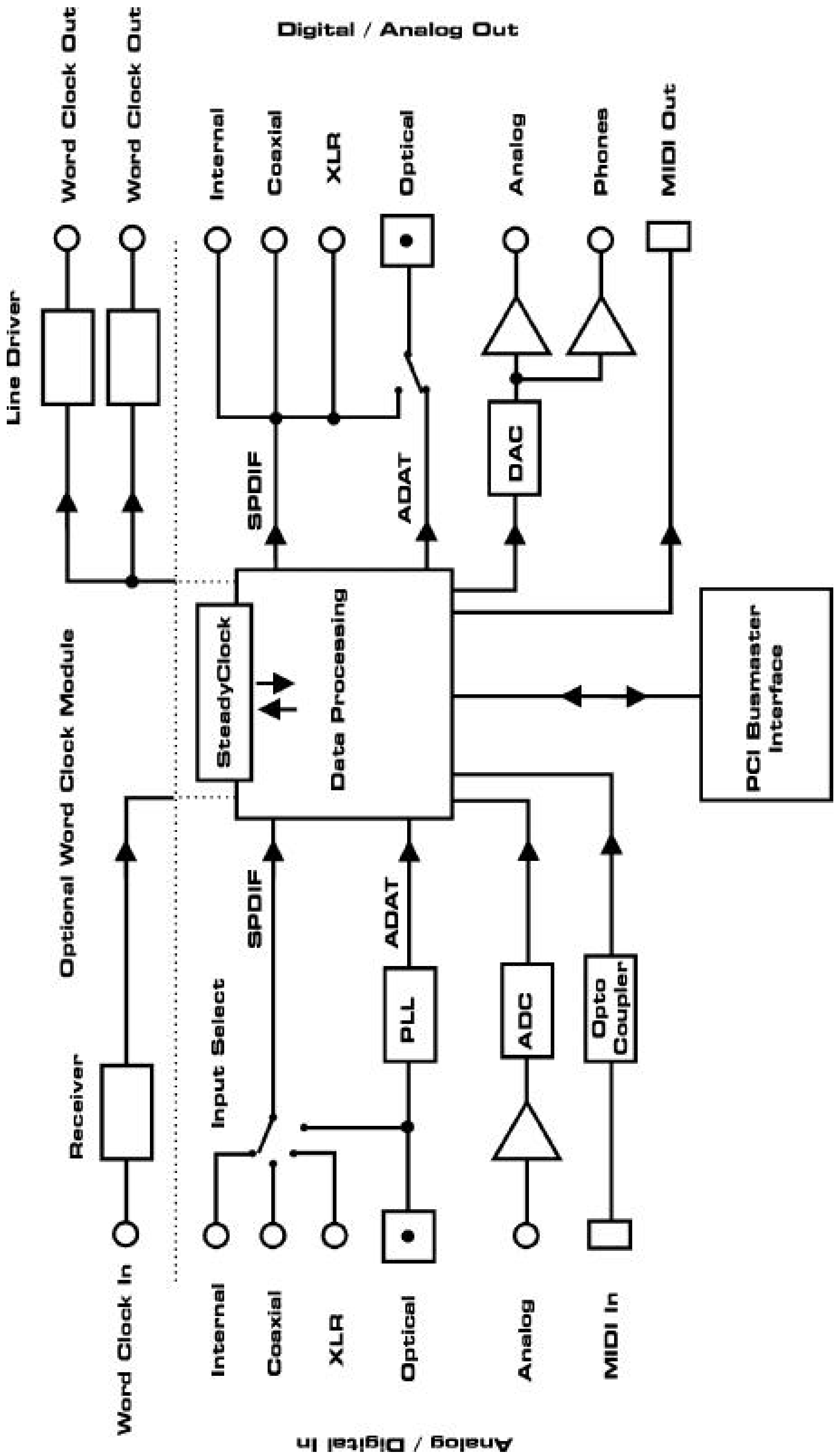
Current driver version: W2k: 2.62

25. Diagrams

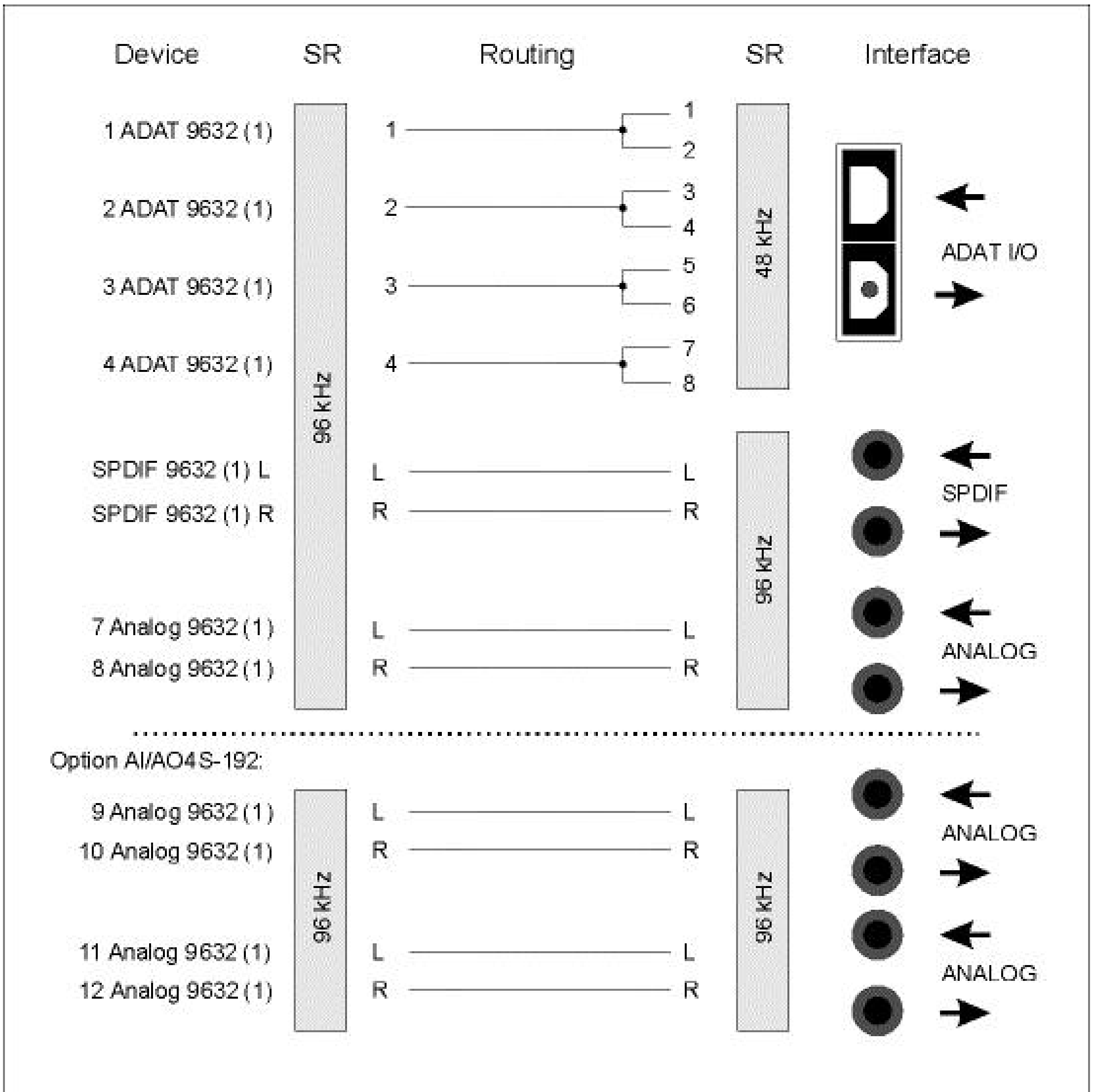
25.1 Block Diagram HDSP 9632



25.2 Channel 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 so that they equal the physical 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, interface, HDSP 9632, card number

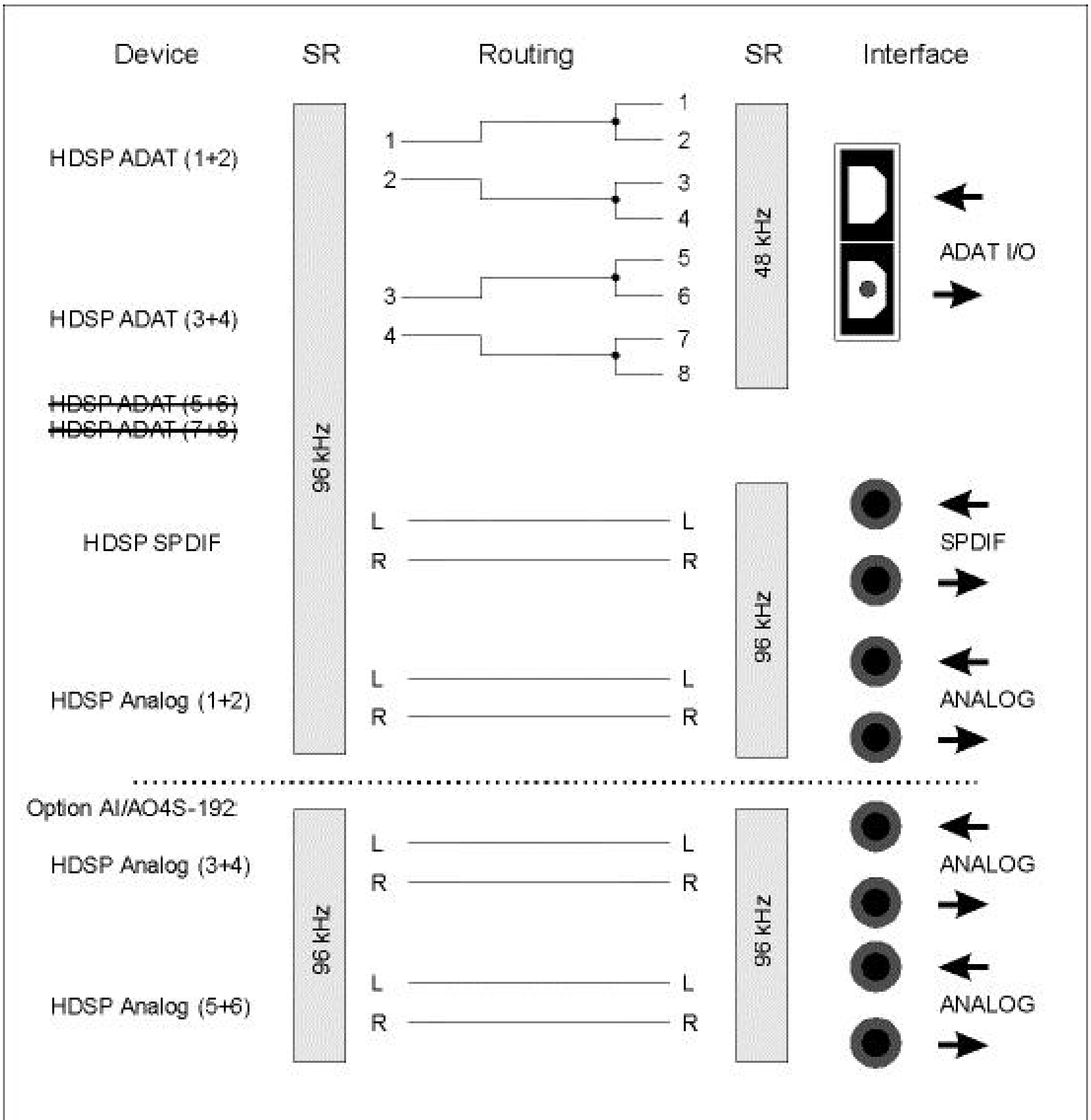


96 kHz Routing HDSP 9632 ASIO

25.3 Channel 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 the 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 9632 MME

25.4 Pin Assignment D-sub Connectors

The 9-pin D-sub connector for the digital breakout cable is identical to the one used in the DIGI96 series (so the cable is identical too).

Pin assignment of the 9-pin D-type connector

Pin	Name	Pin	Name	Pin	Name
1	GND	4	AES Out +	7	SPDIF In -
2	SPDIF Out +	5	AES In +	8	AES Out -
3	SPDIF In +	6	SPDIF Out -	9	AES In -

The 15-pin D-sub connector for the analog breakout cable can be used with an unbalanced (RCA) and balanced (XLR) breakout cable.

Pin assignment of the 15-pin D-type connector and analog XLR breakout cable

Pin	Name	Pin	Name	Pin	Name
1	Line In Left -	6	Line In Left +	11	Line In Right -
2	Line In Right +	7	Line Out Left -	12	Line Out Left +
3	Line Out Right +	8	Line Out Right -	13	Phones Left
4	MIDI Out (5)	9	GND/Shell	14	Phones Right
5	MIDI In (4)	10	MIDI In (5)	15	MIDI Out (4)

Pin assignment of the analog RCA/phono breakout cable

Pin	Name	Pin	Name	Pin	Name
1	GND/Shell	6	Line In Left +	11	GND/Shell
2	Line In Right +	7	n.c.	12	Line Out Left +
3	Line Out Right +	8	n.c.	13	Phones Left
4	MIDI Out (5)	9	GND/Shell	14	Phones Right
5	MIDI In (4)	10	MIDI In (5)	15	MIDI Out (4)

26. 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.