# User's Guide



INTELLIGENT AUDIO SOLUTIONS

# Hammerfall® DSP System

# RPM

**DVD**

TotalMix™

**ASIO**

**24 Bit / 96 kHz ✓ ready**

**SyncAlign®**          **ZLM®**

Mobile Turntable / CD / MIC / MIDI Interface
PCI Busmaster Digital I/O System
PCI and CardBus Interface
24 Bit / 96 kHz

# Contents

## 1. Introduction

Thank you for choosing the RPM. This unique audio system is the perfect tool for DJs, home recording and professional entertainers. It connects microphone, turntable, CD player and other analog sources directly to a computer. Installation is simple, even for the inexperienced user, thanks to the latest Plug and Play technology and full interrupt-sharing. The numerous unique features and well thought-out configuration dialog puts the Hammerfall DSP at the very top of the range of digital audio interface cards.

The package includes drivers for Windows 2000/XP and Mac OS X. An ALSA driver for Linux will soon be 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:

PCI Interface:
* PCI card HDSP
* Quick Info guide
* RME Driver CD
* Cable IEEE1394, 4.5 m (15 ft)
* Internal cable (3 pin)

CardBus Interface:
* CardBus card
* Quick Info guide
* RME Driver CD
* Cable CardBus to IEEE1394, 4.5 m  (15 ft)
* 12 V car cable
* Battery cable
* Power supply 12 V / 1.25 A and power cord

RPM:
* I/O-box RPM
* Quick Info guide
* RME Driver CD


## 3. System Requirements

* Windows 2000/XP, Mac OS 10.2.8 or up
* PCI Interface: a free PCI rev. 2.1 Busmaster slot
* CardBus Interface: a free PCMCIA Slot type II, CardBus-compatible

Note: Examples and detailed descriptions of suitable audio desktop systems can be found in the Tech Info *RME Reference PCs: Hardware recommendations*. Information on compatibility and performance of notebooks/laptops is included in RMEs Tech Infos about notebooks, *HDSP System – Notebook Basics and Tests.*

## 4. Brief Description and Characteristics

- 2 high quality phono inputs with RIAA compensation
- 1 microphone input with Class-A frontend
- Level adjustment in hardware ensures highest dynamic ratio
- All settings can be changed in real-time
- 8 available buffer sizes/latencies: 1.5 / 3 / 6 / 12 / 23 / 46 / 93 / 186 ms
- 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
- 1 x MIDI I/O, 16 channels high-speed MIDI
- 1 x Analog Line/headphone output, separate output for independent submix
- DIGICheck DSP: Level meter in hardware, peak- and RMS calculation
- TotalMix: 102 channel mixer with 40 bit internal resolution


## 5. Technical Specifications

### 5.1 Digital

- Internal sample rates: 32 / 44.1 / 48 / 88.2 / 96 kHz
- Internal resolution: 24 Bit


### 5.2 Analog

**AD Phono Input**
- Resolution AD: 24 Bit
- Frequency response (RIAA), ± 1 dB: 40 Hz - 20 kHz
- Maximum input level (0 dBFS), 1 kHz: -23 dBu, 54 mV
- Signal to Noise Ratio (SNR): 87 dB RMS unweighted, 96 dBA
- THD: < -93 dB, < 0.002 %
- THD+N: < -83 dB, < 0.007 %
- Channel separation @ 1 kHz: > 100 dB
- Input impedance: 30 kOhm
- Input capacity: 220 pF

**AD Line Input**
- Resolution AD: 24 Bit
- Frequency response AD @ 44.1 kHz, -0.5 dB: 5 Hz - 20.7 kHz
- Frequency response AD @ 96 kHz, -0.5 dB: 5 Hz - 32 kHz
- Signal to Noise ratio: 10 dB RMS unweighted, 106 dBA
- THD: < -100 dB, < 0.001 %
- THD+N: < -93 dB, < 0.002 %
- Channel separation @ 1 kHz: > 100 dB
- Input: RCA, unbalanced
- Input impedance @ Line 0 dB: 30 kOhm
- Input impedance @ Line -6 dB: 2 kOhm
- Input level for 0 dBFS @ Line 0 dB: +2.5 dBu, 1 V RMS
- Input level for 0 dBFS @ Line –6 dB: +8.5 dBu, 2 V RMS

**AD Mic Input**

- Resolution AD: 24 Bit
- Frequency response AD @ 44.1 kHz, -0.5 dB: 5 Hz – 20.7 kHz
- Frequency response AD @ 96 kHz, -0.5 dB: 5 Hz - 32 kHz
- Signal to Noise ratio (SNR): 100 dB RMS unweighted, 106 dBA
- THD: < -100 dB, < 0.001 %
- THD+N: < -93 dB, < 0.002 %
- Input: 6.3 mm TRS (stereo) / XLR combo jack, servo-balanced
- Input impedance: 2 kOhm

**DA**

- Resolution DA: 24 Bit
- Signal to Noise ratio: 108 dB RMS unweighted, 111 dBA (unmuted)
- Frequency response DA @ 44.1 kHz, -0.5 dB: 5 Hz – 20.9 kHz
- Frequency response DA @ 96 kHz, -0.5 dB: 5 Hz - 35 kHz
- THD: < - 98 dB, < 0.0013 %
- THD+N: < -91 dB, < 0.002 %
- Channel separation: > 100 dB
- Maximum output level RCA unbalanced: +8 dBu, 2 V RMS
- Maximum output level TRS balanced: +14 dBu, 4 V RMS
- Output impedance: 220 Ohm

**Phones**

- Resolution DA: 24 Bit
- Frequency response DA, -0.1 dB: 20 Hz - 20.8 kHz (sf 44,1 kHz)
- Frequency response DA, -0.5 dB: 10 Hz - 44 kHz (sf 96 kHz)
- Signal to Noise Ratio (SNR): 108 dB (RMS unweighted, unmuted), 111 dBA
- THD: < - 98 dB, < 0.0013 %
- THD+N: < -91 dB, < 0.002 %
- Channel separation: > 100 dB
- Maximum output level: +8 dBu
- Ouput impedance: 30 Ohm

## 5.3 MIDI

- 1 x MIDI I/O via 5-pin DIN sockets
- PCI bus based hi-speed operation
- Separate 128 byte FIFO for input and output
- MIDI state machine in hardware for reduced interrupt request load

## 5.4 Power supply

- The CardBus card does not provide power to the RPM. Therefore a hi-tech switching power supply is included
- The PCI card operates as power supply for the attached RPM via the FireWire cable

The RPM draws a high startup current of more than 2 A during initialisation. Current at 12 Volt operating voltage: unloaded 250 mA (3 Watts), loaded 560 mA (7 Watts). Supply voltage range DC 10 V – 15 V. The center pin of the AUX jack is Plus. The unit includes protection against reversed polarity of the power supply.

## 6. Hardware Installation

### 6.1 PCI Interface

*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 card from its protective bag, discharge any static in your body by touching the metal chassis of the PC.
4.  Insert the PCI card firmly into a free PCI slot, press and fasten the screw.
5.  Replace the computer's housing.
6.  Reconnect all cables including the power cord.
7.  Connect PCI interface and RPM using the supplied cable (IEEE1394). This is a standard FireWire cable (6-pin).

### 6.2 CardBus Card

Before inserting the CardBus card make sure the complete HDSP system is ready for operation!

1.  Connect the CardBus card with the RPM using the supplied cable.
2.  Insert the CardBus card with the Hammer logo up into a PCMCIA slot.
3.  Plug the power jack of the supplied switching power supply into the connector labeled AUX, on the rear of the RPM.
4.  Connect power cord to power supply, plug into AC outlet. The green LED of the power supply and the red LED of the RPM will light up.
5.  Switch on the notebook and boot the operating system.

*The small 15-pin connector of the CardBus card is coded. Only the supplied special cable can be plugged in, and only when the metal sleeve is up. Any kind of violence when plugging in and out can cause damage to the CardBus card.*

## 7. Driver Installation

### 7.1 Windows 2000/XP

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

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

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

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

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

### 7.2 Driver Update under W2k/XP

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

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

## 7.3 Flash Update

The Flash Update Tool updates HDSP PCI cards or CardBus cards to the latest version. It requires an already installed driver.

Start the program hdsp_fut.exe. The Flash Update Tool displays the current revision of the HDSP interface, and whether it needs an update or not. If so, then please manually select if a PCI card (desktop computer) or a CardBus card (laptop) shall be flashed. Next simply press the 'Update' button. A progress bar will indicate when the flash process is finished. The bar moves slowly first (program), then faster (verify).

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

After the update the PCI/CardBus cards need to be resettet. This is done by powering down and shutting off the PC. A warm boot is not enough!

PCI card revision 1.8 or up (black PCB): 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.

All other PCI cards and all CardBus cards: When the update fails (status: failure) the flash process should be repeated several times, until no error message occurs anymore. If the failure message is displayed nonetheless, the interface will most propably no longer work when the computer is switched off and on again. The interface then has to be re-programmed at the factory. We have invested a lot of work to prevent the system from getting in this state. If it happens despite our efforts, the best advice we can give is to not switch off the computer! As long as it is not switched off the old programming of the PCI/CardBus interface will stay active, and you can continue to work with the system using the old drivers

Note: 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 Tray Tools', or 'RME Hammerfall DSP'.

## 7.5 Linux/Unix

A Linux ALSA driver for the Hammerfall DSP system is available. EvenTotalMix has been ported to Linux. Support for the RPM is expected soon. Further information on ALSA is available at

http://www.alsa-project.org

# 8. Operation and Usage

## 8.1 Connections

The front of the I/O-box RPM has the microphone input, MIDI input and output, the headphone output, and several status LEDs:

**Microphone In** allows to connect a microphone via XLR or 1/4" TRS jack.

Using the **GAIN** knob the sensitivity is adjusted to match the currently used microphone.

The **CLIP** LED visualizes overload, caused by too high input level or wrong Gain adjustment. The LED lights up 4 dB before reaching the highest possible (undistorted) level.

**MIDI IN** and **OUT** represent the MIDI input and output, realized as 5-pin DIN jacks.

**MIDI State** indicates sent or received data for the MIDI port

The red **HOST** LED lights up constantly when the power supply or the computer is switched on, thus signalling the presence of operating voltage. At the same time it operates as Error LED, in case the I/O-box wasn't initialised, or the connection to the interface has been interrupted (Error, cable not connected etc.). The latter causes the LED to flash. After the firmware has been loaded into the I/O-box the LED extinguishes, thus signalling an error-free operation.

**Phones** is a low impedance line output of highest quality, which can produce a sufficient and undistorted volume even when used with low-impedance headphones.

The volume for the headphones is controlled with the **VOL** knob.

The back of the RPM has the 4 analog inputs and outputs and the power supply connector **AUX** (only needed in CardBus operation).

The **PHONO/LINE INPUTS 1/2** and **3/4** can be used with turntables (moving magnet) or Line-based devices (CD player etc.). Operation is controlled in the Settings dialog.

**OUTPUTS 1/2** are provided as both RCA jacks (unbalanced) and as stereo TRS jacks (balanced).

**OUTPUTS 3/4** are only available as RCA jacks (unbalanced).

The **hook** serves as strain relief. Originally only intended to secure the power supply cable (simply tie it around the hook, or tie a knot around it), the hook is big enough to hold several of the cables connected to the RPM.

## 8.2 Playback

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

In the audio application being used, 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 below shows a typical configuration dialog as displayed by a (stereo) wave editor. Audio data is sent to the chosen device, which can be one of 6 stereo outputs. 3 of these outputs are virtual and will be mixed by TotalMix to any of the existing 3 hardware outputs.

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'.
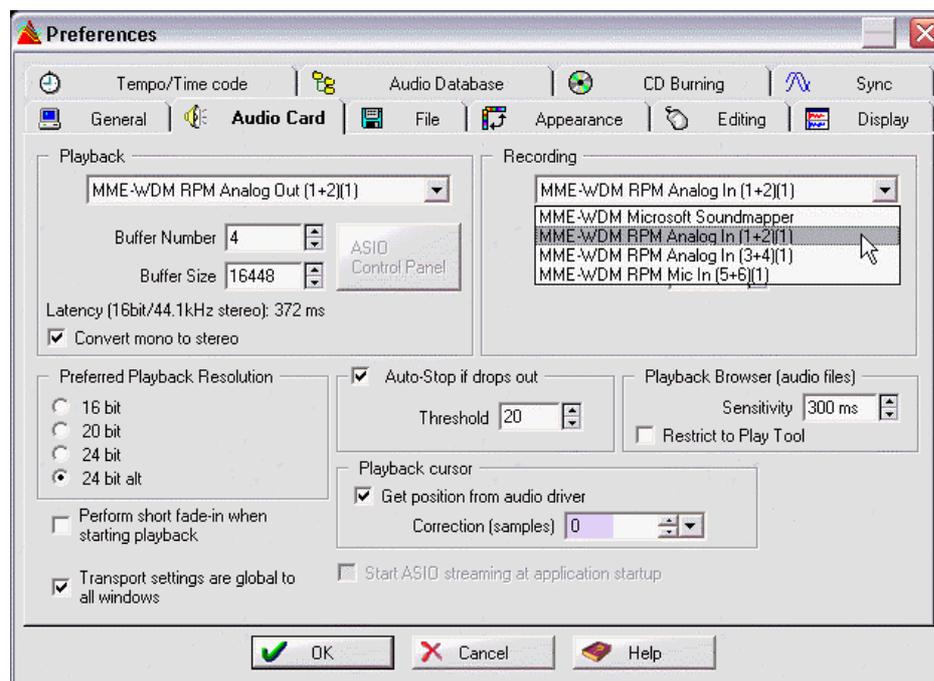
## 8.3 Multi-Client Operation

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. Interesting, but with limited real-world meaning: all inputs can be used simultaneously.

### 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. Again 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.

### Virtual Outputs
The RPM has 6 physical outputs, but the driver provides 12 playback channels, which can be routed and mixed with no limitations to any of the 6 physical outputs. This technique allows for a simultaneuos usage of multiple programs (multi-client operation). Each program uses its own playback channels exclusively. As a result, up to 6 programs can be used at the same time for playback purposes.

## 8.4 DVD-Playback under MME

DVD software player like WinDVD and PowerDVD can operate as software decoder, sending a DVD's multichannel data stream directly to the analog outputs of the RPM. Supported are all modes, from 2 to 8 channels, at 16 bit resolution and 48 kHz sample rate.

For this to work an output wave device of the HDSP 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 RPM.

The device selected as *Preferred Playback Device* defines the first playback channel. Choosing RPM Analog Out 3/4 and 6-channel mode, playback will happen on channels 3 to 8.

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: PowerDVD 5 no longer supports a starting channel other than channel 1.

Note 3: In WinDVD 5, a channel-seperated 5.1 playback using the HDSP requires a change in the registry. Start regedit, go to *HKEY_CURRENT_USER / Software / InterVideo / DVD5 / AUDIOCHAN*, and set its value to *4* (hexadecimal). Additionally select 'Waveout' as Audio Renderer in the Audio configuration dialog. **Warning: Changes to the registry are done at your own risk. Danger of complete data loss!**


## 8.5 Low Latency under MME (Buffer Size Adjustment)

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

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

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

Example: when you set the RPM 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 older VIA based ones) 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 Recording

For recordings via the analog inputs the corresponding record device has to be chosen (RPM Analog (x+x)). The sensitivity of the phono and line inputs can be adjusted coarse in the Settings dialog. In real world operation, a further fine-adjustment prooves not to be necessary. Using line sources, the analog level can usually be controlled directly at the source to match the RPM's sensitivity perfectly.

The sample frequency shown in the Settings dialog (see chapter 9, screenshot Settings) is useful as a quick display of the RPM's current configuration.

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

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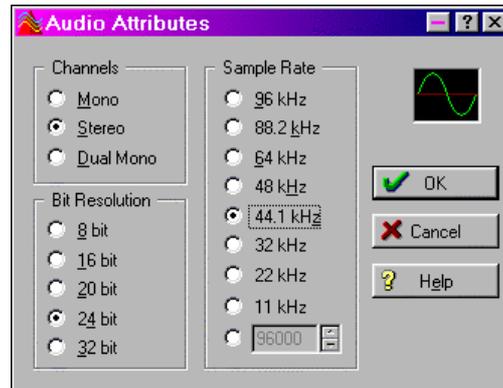
It often makes sense to monitor the input signal or send it directly to the output. This can be done at zero latency using **TotalMix** (see chapter 15).

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

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

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

## 8.7 Analog Inputs

One of the main issues when working with an AD-converter is to maintain the full dynamic range within the best operating level. The RPM includes a discreet hardware level adjustment which allows for a perfect adaptation of both line and phono inputs. While this setting is controlled by software, it operates at hardware level. This way, a maximum dynamic range and sound quality can be guaranteed.

The following input options are avilable:

- Phono +6 dB: amplifies the phono signal by 6 dB. Maintains a sufficient level  for the AD-converter even with low level cartridges and low volume longplay records.

- Phono 0 dB: default sensitivity, matches most cartridges and longplay records

- Phono –6 dB: loweres the phono signal by 6 dB. May be necessary when cartridges or longplay records provide extreme levels.

- Line 0 dB: default sensitivity, matches most tape decks

- Line –6 dB: lowers the Line signal by 6 dB. Matches most CD-players and other line sources.

In case the analog line input is overloaded despite a setting of –6 dB (see Level Meters of TotalMix), the output signal of the connected device must be lowered. When the device's line outputs do not provide output level adjustment, a useful alternative can be to use its headphone output, as these can be adjusted nearly always. A simple cable adapter stereo TRS jack to 2 x  RCA is available in many stores (see next page).

The phono inputs not only heavily amplify the very low level signal of the cartridge, but also correct the frequency response according to RIAA. Bass will be raised, treble will be reduced significantly. Overall a linear frequency response is achieved. Due to the very low input signal, the high amplification and the heavy bass boost, phono inputs do not show Signal to Noise ratios and THD values as known from Line inputs. At first view a SNR of 85 dB seems to be terrible, as even 16 bit of dynamic range (96 dB) aren't achieved. This value has to be put into perspective, which happens the moment you put the needle on the longplay record. Even a 'perfect' empty record produces an unavoidable noise of about –60 dB, far away from the quality of the RPM's phono circuitry and AD-converters.

The RPM includes a fixed first order high-pass filter, also known as rumble filter (although such filters never removed the typical rumble noise, as those 'needle grinding sounds' are found far above 150 Hz). The corner frequency of 35 Hz is very low, preventing audible sound changes. However while testing the RPM this filter has been found to be very useful to remove low frequency problems. Such signals are generated not only when scratching, but also by simple mechanical feedback, or by the uneven longplay record itself.

## 8.8 Analog Outputs

All the RPM's outputs are short circuit protected and low impedance types.

The outputs 1/2 are provided as both RCA jacks (unbalanced) and as stereo TRS jacks (balanced). The electronic output stage does not operate servo-balanced! A connection of unbalanced devices using monaural TS-jacks should be avoided, as power consumption and distortion of the RPM may slightly rise. However, the unit won't be damaged by this.

The maximum output level at the RCA jacks is +8 dBu, equalling 2 V RMS. With this, the RPM is compatible to both Hi-Fi units and professional studio equipment. Lowering the output level by TotalMix up to –10 dB, to match the inputs of lower level devices, does not cause a reduced playback quality, because the DA-converters offer lots of dynamic range headroom by their Signal to Noise of about 110 dB.

The maximum output level at the TRS-jacks is +14 dBu, equalling 4 V RMS. Thanks to this high-level balanced output, the RPM can be connected to mixing desks, stage boxes, well even directly to power amplifiers.

The RCA outputs 3/4 have the same technical data as outputs 1/2.

The headphone output at the front uses a doubled driver stage, to provide sufficient volume even with low-impedance headphones. All other techical data equal the ones of the RCA outputs 1/2/3/4. The VOL pot allows a stepless adjustment of the output level directly at the unit.

The headphone output can also be used as additional hi-quality line output. A simple a simple cable adapter stereo TRS-jack to 2 x RCA is needed, see picture to the right, is required then.

The pin assignment follows international standards. The left channel is connected to the tip, the right channel to the ring of the TRS jack/plug.
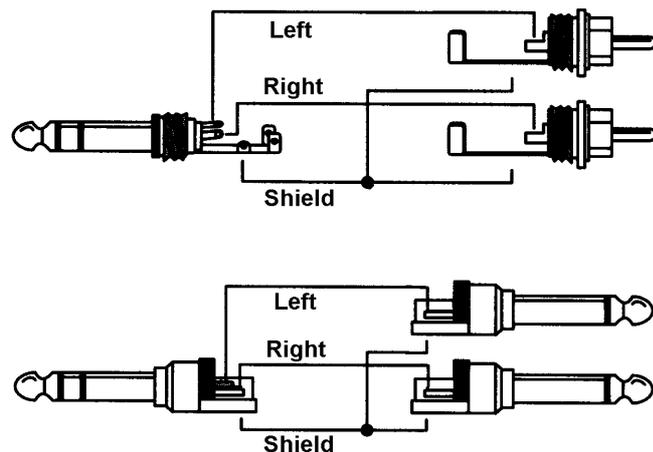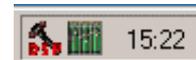
# 9. Configuring the RPM
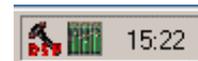
## 9.1 General Information

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

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



The mixer of the Hammerfall DSP System (TotalMix) can be opened in two different ways:

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



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

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.

The status displays at the bottom of the dialog box give the user precise information about the current status of the system.
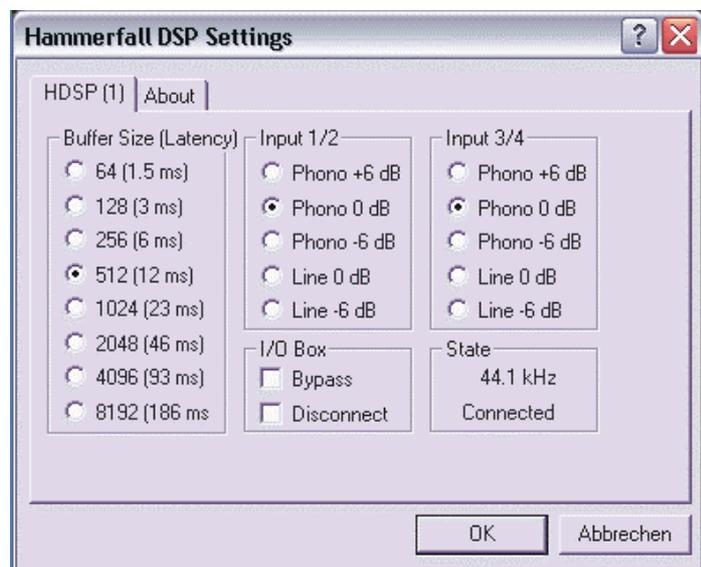


**Buffer Size**
The setting *Buffer Size* determines the latency between incoming and outgoing ASIO and GSIF data, as well as affecting system stability. Under Windows MME this setting determines the DMA buffer size (see chapter 8.5).

**Input 1/2**
Changes between phono and Line input and several sensitivities. Setting *+6 dB* amplifies the input signal by 6 dB, *–6 dB* loweres it by 6 dB.

**Input 3/4**
As Input 1/2.

**I/O Box**
*Bypass* acitvates a hardware bypass, which connects the phono preamps output signals directly to the outputs 1/2 and 3/4. A playback of other signals is not possible, neither by the computer nor by TotalMix. The headphone output will operate as usual.

*Disconnect* interrupts the communication between I/O-box and PCI or CardBus card. In case the RPM has been configured using the Settings dialog and TotalMix, it can be used Stand-Alone (without a connected computer) after activation of *Disconnect* and after a power supply has been attached.

**State**
This display shows the current sample rate and state of the I/O-box:

*Error*: I/O-box not connected or missing power

*Detected*: The interface has found an I/O-box and tries to load the firmware

*Connected*: Communication between interface and I/O-box operates correctly

*Disconnected*: Communication between interface and I/O-box has been interrupted, I/O-box continues operation
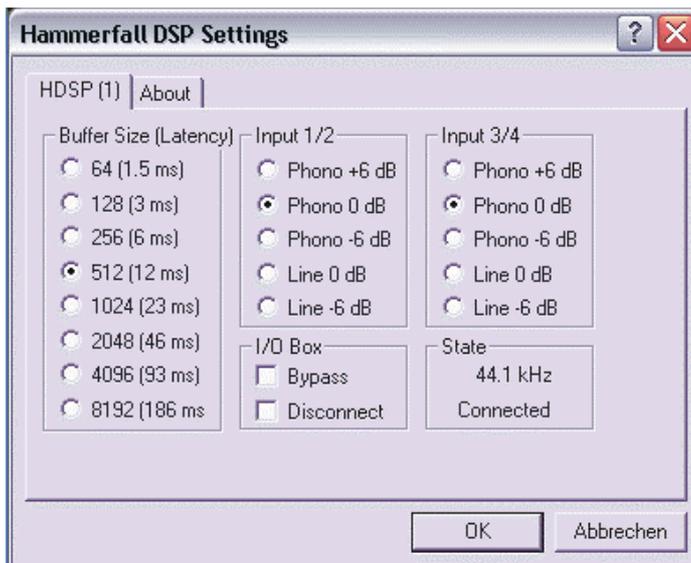
## 10. Disconnect Mode

RME's exclusive Disconnect mode lets you adjust level, input selection and signal mix via your computer, and simply detach the RPM afterwards. With this, a stand-alone operation of the RPM gets possible. When the RPM has been configured using Settingsdialog and TotalMix, it won't loose those settings after detaching it from the computer. Provided, the RPM is operated with an external power supply, because the RPM (as all HDSP I/O-boxes) does not contain memory, so alll settings get lost upon power-off.

If you do not need to record or playback audio with your computer, the RPM can be used in a fixed configuration, for example as:

• Phono preamp with monitoring
• Phono and Line preamp with mixing functionality
• Microphone preamp
• Headphone mixer
• Line splitter wioth balanced output
• what ever you just need!

The unit can also stay connected to a desktop computer, receiving its power from there. The Disconnect mode will then cause the unit to be isolated from the computer, no longer be available to any software or other changes – an interesting security application.

## 11. Bypass Mode

Directly after power-on, in case of power failure or with your computer disconnected, the RPM is in *Bypass* mode. The phono inputs are set to 0 dB, the phono preamps output signal is hard-patched to the Line outputs by means of a relais. Even in an emergency the unit will still serve as a phono preamp and prevent a complete audio dropout.

The *Bypass* mode activated in the Settings dialog only activates the relais, all other settings remain valid (level/Line etc.). As can be seen in the block diagram (chapter 25), a playback of other signals than the phono inputs is not possible, neither by the computer nor by TotalMix. But the headphone output will continue with normal operation.

## 12. Using more than one Hammerfall DSP

The drivers support multiple interfaces and any combination of I/O-boxes. As the RPM does not include digital interfaces – and therefore no synchronization capabilities - an operation of multiple RPM will lead to a slow drift of the units. Each RPM uses an internal quartz as time reference. These quartz are not completely identical. A long-term playback or a recording on multiple tracks will have deviations in form of small position differences (timing drift).

The RPM can be synchronized to MTC or MIDI clock. In both cases, the used software is repsonsible for proper synchronization to other devices. In case a project includes audio tracks, the software must be capable of recalculating the audio data in realtime to the speed of the MIDI data (Sample Rate Conversion, resampling).

## 13. Operation under ASIO 2.0

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

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

Hammerfall DSP supports 'ASIO Direct Monitoring' (ADM). Please note that some programs do not support ADM correctly.
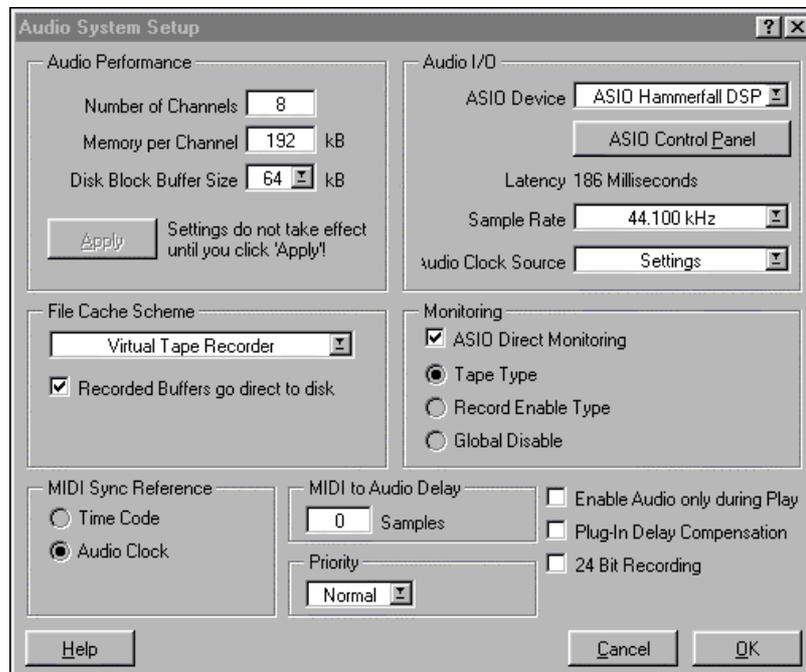
## 14. Operation under GSIF (Gigasampler Interface)

The GSIF interface of the Hammerfall DSP allows direct operation with Gigastudio, with up to 12 channels, 96 kHz and 24 bit.

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

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

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

Thanks to full compatibility to GSIF 2.0, the RPM can also be used as recording device for Gigastudio 3.0.


**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 RPM shows that the record signal always stays un-altered, but can be passed on as often as desired, even with different levels. The level meter of inputs and playback channels are connected pre-fader (due to the enormous routing capabilities). The level meters of the hardware's outputs are connected post-fader.



RME RPM TotalMix
Hammerfall DSP Mixer
Block Diagram, 04/2004
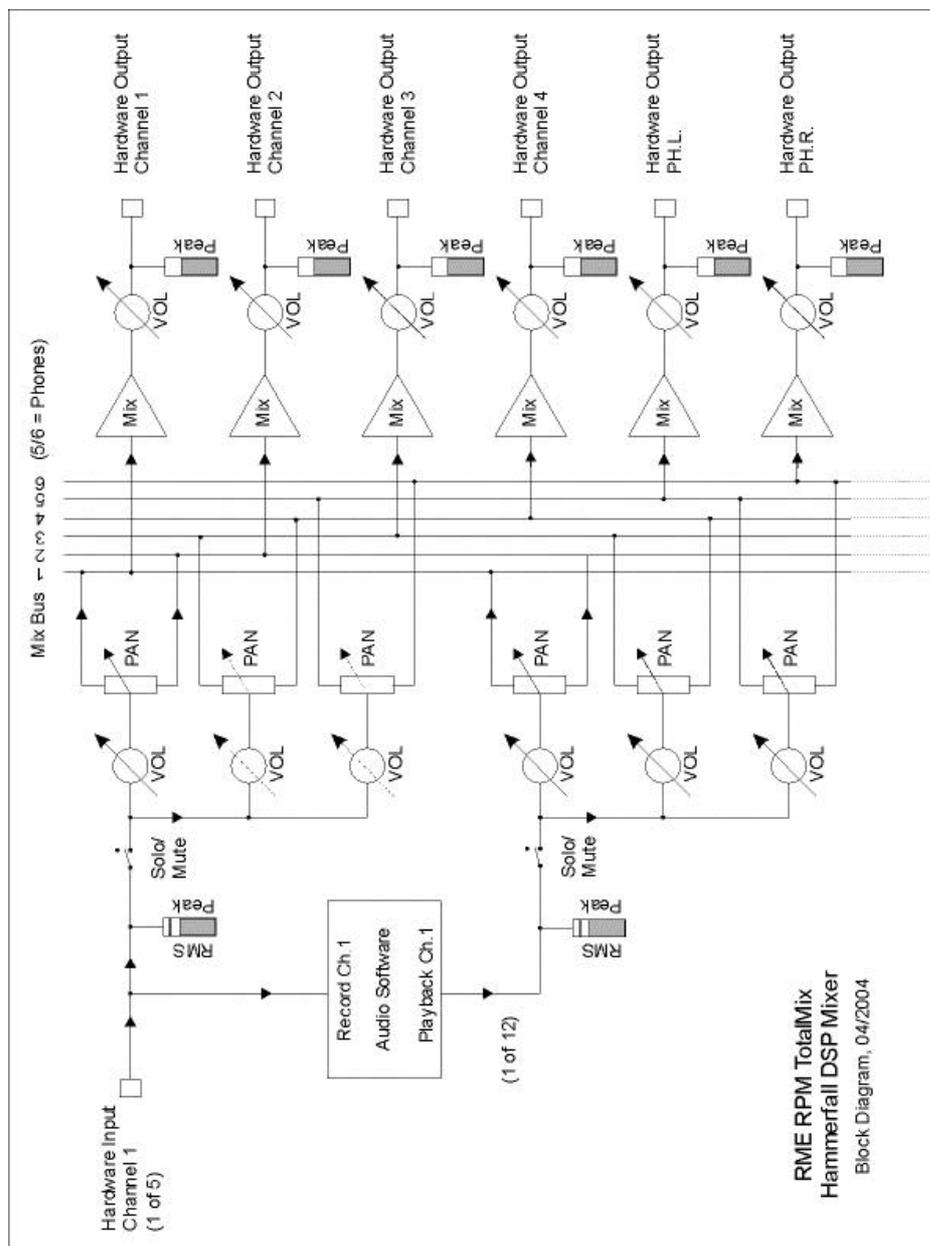
## 15.1 Elements of the Surface

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

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

Every single channel has various elements:

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

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

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

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

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

The white area shows the channel name, the black area shows the current routing target. Selecting one or more channels is done by clicking on the white label which turns yellow then.

## 15.2 Tour de TotalMix

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

We will now create a small submix for the headphone output. Please start a multitrack playback and connect your headphones to the headphone 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 'AN 1+2'. Click onto 'Phones'. The list disappears, the routing window no longer shows 'AN 1+2', but 'Phones'. 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 5 and 6 (PH.L. and PH.R.), you can also see the level of what you are hearing in the phones. The level meter of the hardware output shows the outgoing level. Click into the area above the fader and drag the mouse in order to set the panorama, in this case the routing between channels 5 and 6. 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 headphone 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 'AN 3+4'. Click onto 'Phones'. Now, channel 4 has already been set to 'Phones' as well.

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

Please set 'Out 4' to a gain of around -20 dB and the pan close to center. Now click onto the routing window. You'll now see two checkmarks, one at 'AN 3+4', the other one at 'Phones'. Click onto 'AN 1+2'. The window disappears, fader and panpot jump to their initial values, the signal can now be routed to the analog output 1/2. 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 headphone mix has not changed, while you were routing the channel to other outputs and setting different gain values. With all analog 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 1+2' in the routing window, pull the fader down, open the routing window again - the checkmark is gone.
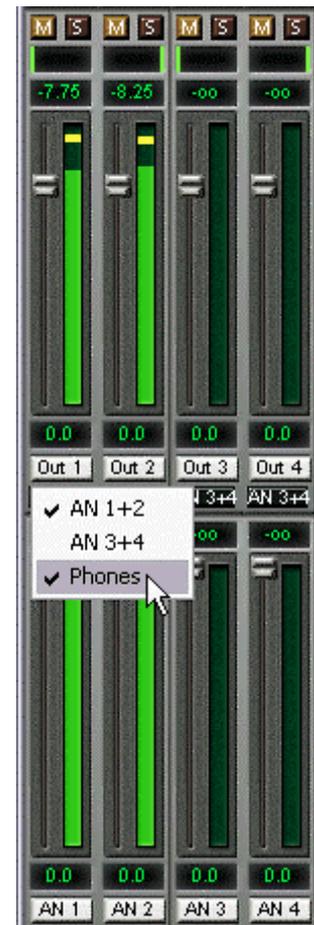
## 15.3 Submix View

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 9 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. So you can then see immediately, which channels, which fader and pan settings make a submix (for example 'Phones').

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 (RPM, Digiface, Multiface, HDSP 9652, HDSP 9632). Holding down Ctrl while clicking on button Card2 will open a second window, instead of replacing the current window content.

## 15.7 Presets

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

This method offers two major advantages:

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

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

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

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

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

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

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

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

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

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

**Preset5.mix**
Description: All faders m.a. As Preset 1, but all Inputs/Playbacks m.a.

**Preset6.mix**
Description: Submix on AN 1+2. As Preset 1, plus submix of all Playbacks on AN 1+2. View Submix AN 1+2 active.

**Preset7.mix**
Description: Submix on AN 1+2. As Preset 6, but all inputs set to 0 dB (1:1 pass through). View Submix 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 (preset4.mix), which can be used to verify and test this functionality. But it is also possible to use any other mixer state, as long as the recording channels are muted manually.

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

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

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

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

## 15.9 Menu Options

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

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

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

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

### 15.10 Level Meter

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

The level meters integrated in TotalMix - considering their size - cannot be compared with the *HDSP Meter Bridge* (chapter 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 RPM will therefore show around -113 dB on all 8 channels.

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

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

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

## 16. The Matrix

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

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

### 16.1 Elements of the Surface

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

- Horizontal labels: All hardware outputs

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

- Green 0.0 dB field. Standard 1:1 routing

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

- Orange gain field. This routing is muted.



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

### 16.2 Operation

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

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

Note the difference between the left side, representing the inputs and software playback sources, and the upper side, representing the hardware outputs. If you move a fader in row 1 or 2 in TotalMix, only the specific levels (max. 2) of this routing will change in the Matrix. But moving a fader in row 3 will make all vertically activated levels move at once (for example 5/6, Phones 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 4 (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 4, open the menu *Edit*. It shows 'Copy In 4'. Now select the desired new input, for example In 1. The menu now shows 'Paste In 4 to In 1'. 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 1/2, but now need the exact same signal also on the outputs 3/4. Click on Out 1, Ctrl-C, click on Out 3, Ctrl-V, same with 2/4 - 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.

## 17.4 Delete Routings

The fastest way to delete complex routings: select a channel in the mixer view, click on the menu entry *Edit* and select *Delete*. Or simply hit the Del-key. Attention: there is no undo in TotalMix, so be careful with this function!

## 18. Notes on using Laptops and CardBus

The HDSP system uses the notebook's PCMCIA type II port as *CardBus* interface. Compared to a PC-Card, which only has access to the outdated ISA-bus, CardBus is a 32 bit PCI interface. Like with a desktop system it's not possible to remove a PCI device while in operation. First the operating system has to receive a ‚removal request', then the device has to be stopped. Finally the card can be pulled out of the PCMCIA slot.

To remove the hardware click on the green arrow symbol in the systray. It is possible to stop the HDSP directly, or to first call up the info dialog by double clicking the symbol, and then stopping it.

When inserting the CardBus card it usually will be detected automatically by the notebook hardware and then Windows. A double beep signals detection and removal. In rare cases detection will fail. If so, simply remove the card and insert it again. On one of the tested notebooks the card was never activated at Windows boot, but only when plugged in manually. The problem vanished after a fresh install of Windows ME.

The Hammerfall DSP System was tested thoroughly on several notebooks by RME. Basically our CardBus card operates with any notebook we have tested. But there were some significant performance differences, and some laptops suffered from a very low data  transfer rate of the PC-Card controller. Detailed information on this can be found in the Tech Info *HDSP System: Notebook Tests – Compatibility and Performance*.

The mobile operation of the HDSP system can cause problems. Explanations and solutions on digital noise, ground loops, headphone operation and Line Out wiring, power supplies and the mobile operation with battery can be found in the Tech Info *HDSP System: Notebook Basics - The Audio Notebook in Practise*.

The hardware of a notebook differs in many points from that of a desktop computer – sometimes…Detailed information on all components, from CPU to the display, can be found in the Tech Info *HDSP System: Notebook Basics – Notebook Hardware*.

Why does your notebook behave completely different than your desktop? And how you can still change it for good performance (not only audio performance)? Find answers to these and many other questions about software tuning and installation, clicks under Win9x, the usage of the low latency solution Windows 2000/XP, and the optimization of the memory bandwidth, in the Tech Info *HDSP System: Notebook Basics – Background Knowledge and Tuning*.

## 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 globally disabled.

*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.

*Crackle during record or playback:*

- Increase the number and size of buffers in the 'Settings' dialog or in the application.
- Try different cables to rule out any defects here.
- 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.

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

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

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

- Check whether the CardBus card is completely inserted into the PCMCIA slot, or 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 4.1

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 4.1 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. K-system meter, free scaling/referencing.
- **Vector Audio Scope**. World wide unique Goniometer showing the typical afterglow of a oscilloscope-tube. Includes Correlation meter and level meter.
- **Spectral Analyser.** World wide unique 10-, 20- or 30-band display in analog bandpass-filter technology. 192 kHz-capable!
- **Bit Statistics & Noise**. Shows the true resolution of audio signals as well as errors and DC offset. Includes Signal to Noise measurement in dB and dBA, plus DC measurement.
- **Totalyser**. Spectral Analyser and Vector Audio Scope as one window function.
- **Channel Status Display**. Detailed analyzis and display of SPDIF and AES/EBU Channel Status data.
- **Completely Multi-client.** Open as many measurement windows as you like, on any channels and inputs or outputs!

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


### 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 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, further increasing the flexibility and usability of the HDSP system. Additionally parts of the HDSP system, like the special CardBus cable and the switching power supply, are available seperately.

**Part Number          Description**
36000                        19", 1UH Universal rack holder

This 19" rack holder has holes for Digiface and RPM. Two units can be installed side by side in any combination. The rack holder also includes holes for nearly all 19" half-rack units from other manufacturers.

36001                        Firewire cable IEE1394 6M/6M, 1 m (3.3 ft)
36002                        Firewire cable IEE1394 6M/6M, 3 m (9.9 ft)
36005                        Firewire cable IEE1394 6M/6M, 5 m (16.4 ft)
36010                        Firewire cable IEE1394 6M/6M, 10 m (32.8 ft)

Firewire cable for the HDSP system, both sides 6-pin male. Cable longer than 16 ft is not allowed for Firewire, therfore hard to get in computer shops. However the HDSP system can operates flawlessly even with a cable length of up to 50ft (15 m).

36081                        RME Firewire cable for CardBus 15/6M, 5 m (16.4 ft)
36082                        RME Firewire cable for CardBus 15/6M, 1 m (3.3 ft)

Special cable 15-pin close Lan coded to 6-pin male, for RME CardBus card.


37011                        Power supply for HDSP CardBus card

Robust and light weigth switching power supply, 100V-240V AC, 12V 1.25 A DC.

## 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.

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 and 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 OHG apply at all times.


## 24. Appendix

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

http://www.rme-audio.com

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


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Current driver version: W2k/XP: 2.7

Although the contents of this User's Guide have been thoroughly checked for errors, RME can not guarantee that it is correct throughout. RME does not accept responsibility for any misleading or incorrect information within this guide. Lending or copying any part of the guide or the RME Driver CD, or any commercial exploitation of these media without express written permission from RME Intelligent Audio Solutions is prohibited. RME reserves the right to change specifications at any time without notice.

## 25. Block Diagram RPM
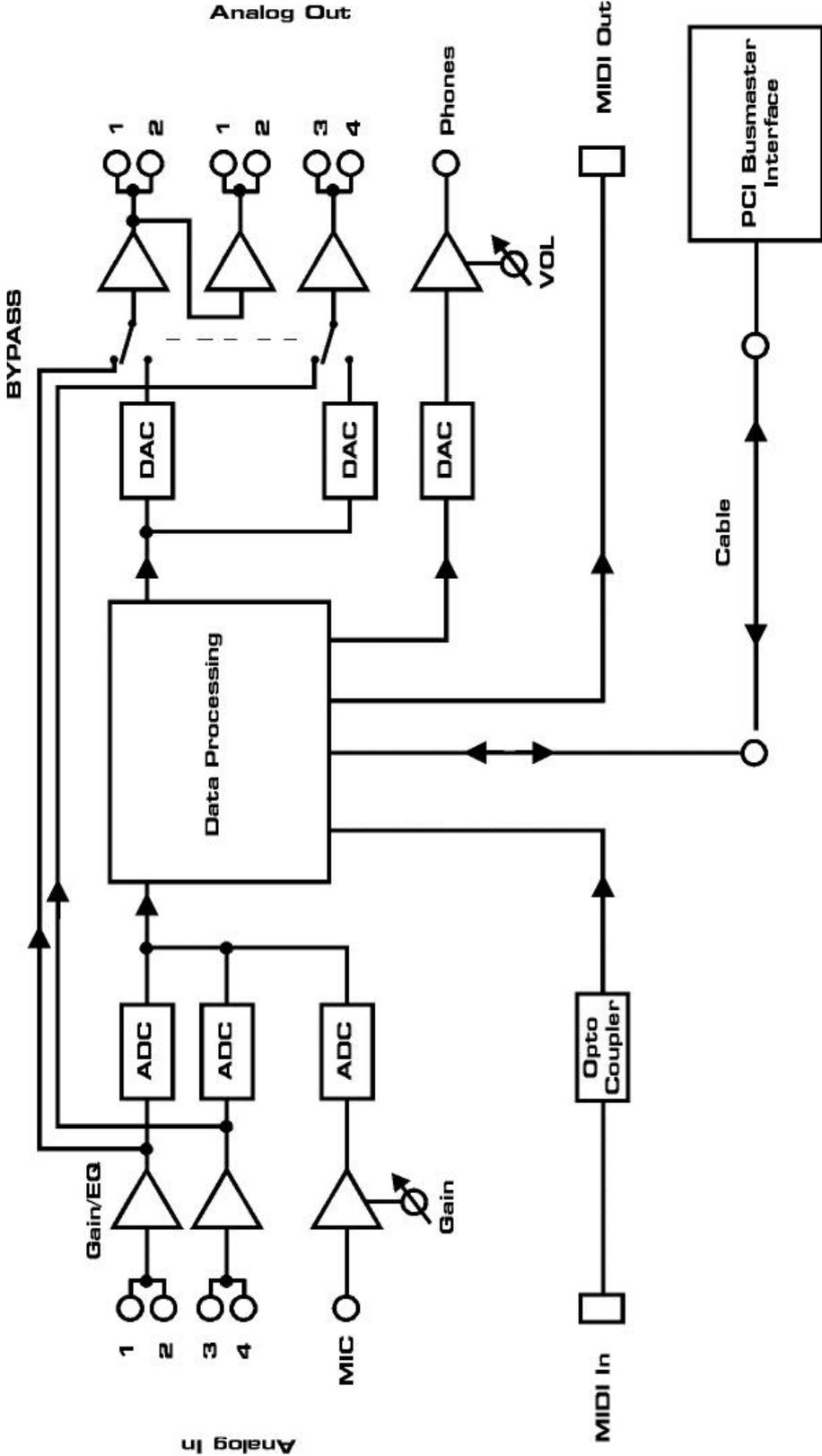
## 26. CE and FCC Compliance Statements

**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.