User's Guide

Hammerfall® DSP System

AES-32

DVD ready 24 Bit / 192 kHz

TotalMix™

ASIO

SyncAlign®

ZLM®

SyncCheck®

SteadyClock™

PCI Busmaster Digital I/O Card
32 Channels AES/EBU Interface
24 Bit / 192 kHz Digital Audio
32 x 16 Matrix Router
2 x MIDI I/O
Quick Boot

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User's Guide

HDSP AES-32

- **General**

1. Introduction

Thank you for choosing the Hammerfall DSP AES-32. This unique audio system is capable of transferring digital audio data directly into a computer, from any device equipped with a AES/EBU or SPDIF interface. Installation is simple, even for the inexperienced user, thanks to the latest Plug and Play technology. The numerous unique features and well thought-out configuration dialog puts the Hammerfall DSP AES-32 at the very top of the range of digital audio interface cards.

The package contains drivers for Windows 2000 SP4, Windows XP, and Mac OS X.

Our high-performance philosophy guarantees maximum system performance by executing as many functions as possible not in the driver (i.e. the CPU), but directly within the audio hardware.

2. Package Contents

Please check your HDSP AES-32 package contains each of the following:

- HDSP AES-32 PCI card
- HDSP AES-32 expansion board
- Quick Info guide
- RME Driver CD
- MIDI breakout cable
- Expansion Board ribbon cable (26-conductor)

3. System Requirements

- Windows 2000 SP4, Windows XP, Mac OS X (10.28 or higher)
- PCI Interface: a free PCI rev. 2.1 Busmaster slot

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
- 8 available buffer sizes/latencies: 1.5 / 3 / 6 / 12 / 23 / 46 / 93 / 186 ms
- 32 channels 192 kHz/24 bit record/playback
- Automatic and intelligent master/slave clock control
- Word clock input and output
- TotalMix for latency-free submixes and perfect ASIO Direct Monitoring
- SyncAlign guarantees sample aligned and never swapping channels
- SyncCheck tests and reports the synchronization status of input signals
- 2 x MIDI I/O, 32 channels high-speed MIDI
- DIGICheck DSP: Level meter in hardware, peak- and RMS calculation
- TotalMix: 512 channel mixer with 42 bit internal resolution
- SteadyClock: Jitter-immune, super-stable digital clock
- Quick Boot technology for immediate loading of the hardware settings

5. Hardware Installation

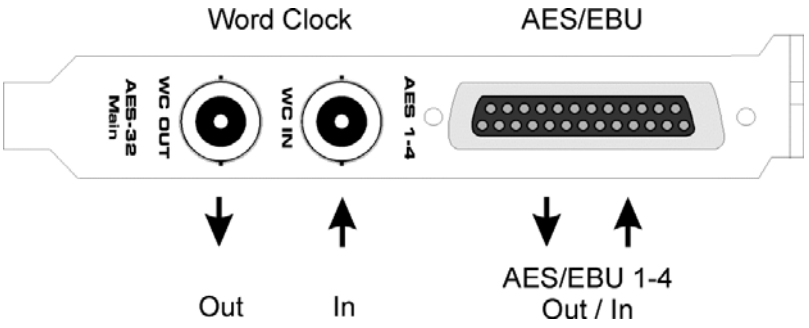
Before installing the PCI card, please make sure the computer is switched off and the power cable is disconnected from the mains supply. Inserting or removing a PCI card while the computer is in operation can cause irreparable damage to both motherboard and card!

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 AES-32 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 AES-32 card to the Expansion Board using the supplied flat ribbon cable.
5. Insert the HDSP AES-32 firmly into a free PCI slot, press and fasten the screw.
6. Insert the Expansion Board and fasten the screw.
7. Replace the computer's housing.
8. Reconnect all cables including the power cord.

6. Hardware - Connectors

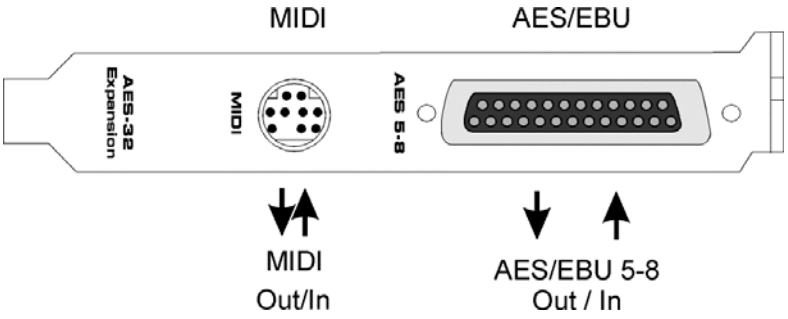
6.1 External Connectors

The bracket of the mainboard has a D-sub 25 pin connector providing **AES/EBU** inputs and outputs **1-4**, and BNC sockets providing **word clock** input and output. The D-sub connector uses the Tascam pinout (details see chapter 30.9).



Breakout and connection cables with this pinout are widely available.

The Expansion Board's bracket has **AES/EBU** inputs and outputs **5-8** via a second D-sub 25 connector. The included breakout cable is connected to the **9-pin Mini-DIN** connector and provides **two MIDI** inputs and outputs via four **5-pin DIN** connectors.



Note: If neither AES I/O 5-8 nor MIDI I/O are required, it is not necessary to install the Expansion Board at all.

Optional TCO
The optional *Time Code Option* is connected to the mainboard with a 10-pin flat ribbon cable. The jack is located right beside the Expansion Board's one. Further details can be found in the TCO's manual.

6.2 Internal Connectors

X100 (Channel 17-32)
26-pin connector for the included HDSP AES-32 Expansion Board.

X101
10-pin connector for a connection of the optional Time Code Option (TCO).

X300
No function. Used to program the card in the factory.

Blue Jumper (Expansion Board)
Controls termination of the word clock input. Changing the jumper to the position oriented towards the center of the PCB (see printed label on the PCB), the word clock input is terminated with 75 Ohms.

7. Accessories

RME offers several optional components. Additionally parts of the HDSP AES-32, like the special breakout cables, are available separately.

Part Number	Description
BOHDSP9652	MIDI breakout cable
VKHDSPAES32	26-conductor flat ribbon cable
BO25MXLR4M4F3	Digital breakout cable AES/EBU, 9.9 ft (3 m)
BO25MXLR4M4F6	Digital breakout cable AES/EBU, 19.8 ft (6 m)
BOBDSUB25T	Digital connection cable 25 pin D-sub, Tascam pinout
BOB32	BOB-32, Universal Breakout Box, 19", incl. 2 x D-sub cable

8. Warranty

Each individual Hammerfall DSP undergoes comprehensive quality control and a complete test at RME before shipping. The usage of high grade components allow us to offer a full two year warranty. We accept a copy of the sales receipt as valid warranty legitimation.

If you suspect that your product 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.

9. 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 browse through a complete copy of the RME website, found on the RME Driver CD (in the **\rmeaudio.web** directory).

Manufacturer:
IMM Elektronik GmbH, Leipziger Strasse 32, D-09648 Mittweida, Germany

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Current driver version: W2k/XP: 2.5, Mac OS X: 2.5

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CE / 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

FCC compliance statement: Tested to comply with FCC standards for home or office use.

User's Guide

HDSP AES-32

Driver Installation and Operation - Windows

10. Driver and Firmware

10.1 Driver Installation

After the HDSP AES-32 has been installed correctly (see 5. 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 **\MADI_AES32_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 AES-32 is ready for use.

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!

10.2 Driver Update

RME's driver updates often include a new madi_aes.inf file. Also the revision number of the hardware might change (after a flash update). To prevent Windows 2000/XP from using an old madi.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 AES-32 /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.

10.3 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 additional autorun entries of TotalMix, the Settings dialog, and the registration 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 MADI and AES-32'.

10.4 Firmware Update

The Flash Update Tool updates the HDSP AES-32 to the latest firmware version. It requires an already installed driver.

Start the program **madi_aes_fut.exe**. The Flash Update Tool displays the current revision of the HDSP AES-32, and whether it needs an update or not. If so, then please 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 card need 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: 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 10.2.

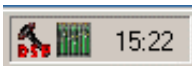
11. Configuring the HDSP AES-32

11.1 Settings Dialog

Configuration of the HDSP AES-32 is done via its own settings dialog. The panel 'Settings' can be opened:

- by clicking on the hammer symbol in the Task Bar's system tray

The mixer of the HDSP AES-32 (TotalMix) can be opened:



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

The hardware of the HDSP system 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
- Configuration of digital I/Os
- Synchronization behaviour
- State of input and output
- Current sample rate
- Latency

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 digital signals.

Quick Boot
All the card's settings described below are stored in a hardware memory, and are loaded immediately after a power-on of the computer. In clock mode Master even the last used sample rate is set. Directly after switching on the computer, a stable and predictable clock state is found at the HDSP AES-32's outputs. This advanced technology completely eliminates disturbing noises and clock network problems during power-up or re-boot.

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

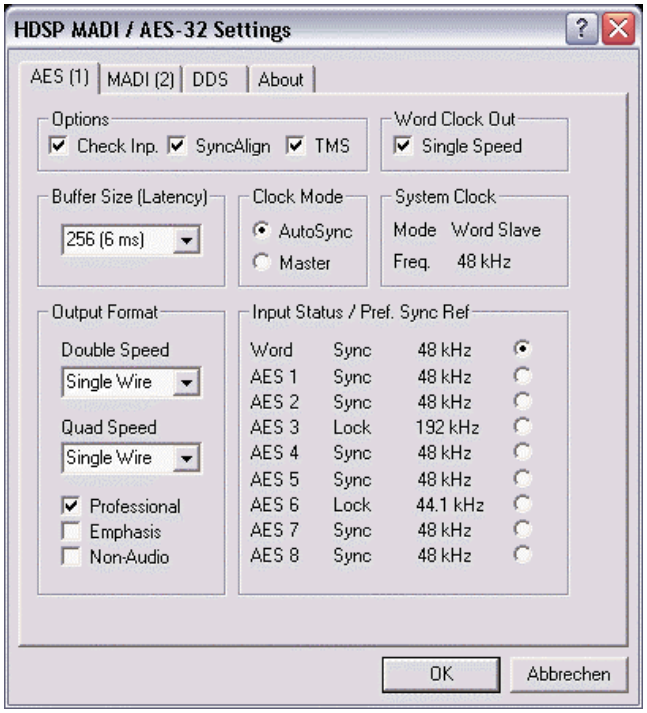
SyncAlign guarantees synchronous channels when using MME multitrack software. This option should only be switched off in case the used software does not work correctly with SyncAlign activated.

TMS activates the transmission of Channel Status data and Track Marker information from the AES/EBU input signal.

Word Clock Out
The word clock output signal usually equals the current sample rate. Selecting *Single Speed* causes the output signal to always stay within the range of 32 kHz to 48 kHz. So at 96 kHz and 192 kHz sample rate, the output word clock is 48 kHz.

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 13/14). Under Windows MME this setting determines the DMA buffer size (see chapter 12.3).

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



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

Output Format
Double Speed: Sample rates in the range of 64 kHz to 96 kHz can also be transmitted using the standard 48K frame. Note that when selecting *Double Wire* the number of output channels is reduced to 8.

Quad Speed: Sample rates in the range of 128 kHz to 192 kHz can also be transmitted using either the standard 48K frame, or the 96K frame. Note that when selecting *Double Wire* the number of output channels is reduced to 8. When selecting *Quad Wire* the number of remaining output channels is 4.

For further details about the settings ‘Professional’, ‘Emphasis’ and ‘Non-Audio’, please refer to chapter 22.2.

Input Status
SyncCheck indicates whether there is a valid signal (Lock) for the inputs Word Clock and AES 1 to 8, or if there is a valid *and* synchronous signal (Sync). The *System Clock Mode* display shows the input and frequency of the current sync source. Each input has its own frequency measurement and display of the input signal's current sample rate.

Pref. Sync Ref.
Used to pre-select the desired clock source. If the selected source isn't available, the unit will change to the next available one. The current clock source and sample rate is displayed in the *System Clock* field.

11.2 Settings dialog - DDS

Usually soundcards and audio interfaces generate their internal clock (master mode) by a quartz. Therefore the internal clock can be set to 44.1 kHz or 48 kHz, but not to a value in between. SteadyClock, RME's sensational Low Jitter Clock System, is based on a *Direct Digital Synthesizer* (DDS). This superior circuitry can generate nearly any frequency with highest precision.

DDS has been implemented into the HDSP AES-32 with regard to the needs of professional video applications, as well as to maximum flexibility. The dialog DDS includes both a list of typical video frequencies (so called pull up/pull down at 0.1% and 4%) and two faders, which allow to freely change the basic sample rate in steps of 1 Hz (!).

Application examples
DDS allows for a simultaneous change of speed and tune during record and playback. From alignment to other sources up to creative effects – everything is possible..

DDS allows to intentionally de-tune the complete DAW. This way, the DAW can match instruments which have a wrong or unchangeable tuning.

DDS allows to define a specific sample rate. This feature can be is useful in case the system randomly changes the sample rate – for unknown reasons. It also prevents a change from Double Speed (96 kHz) to Single Speed (48 kHz), which would cause configuration and routing problems by the changed amount of MADI channels.

The DDS dialog requires the HDSP AES-32 to be in clock mode Master! The frequency setting will only be applied to this one specific card!

Changing the sample rate in bigger steps during record/playback often results in a loss of audio, or brings up warning messages of the audio software. Therefore the desired sample rate should be set at least coarsely before starting the software.

DDS
Activates all settings of this dialog.

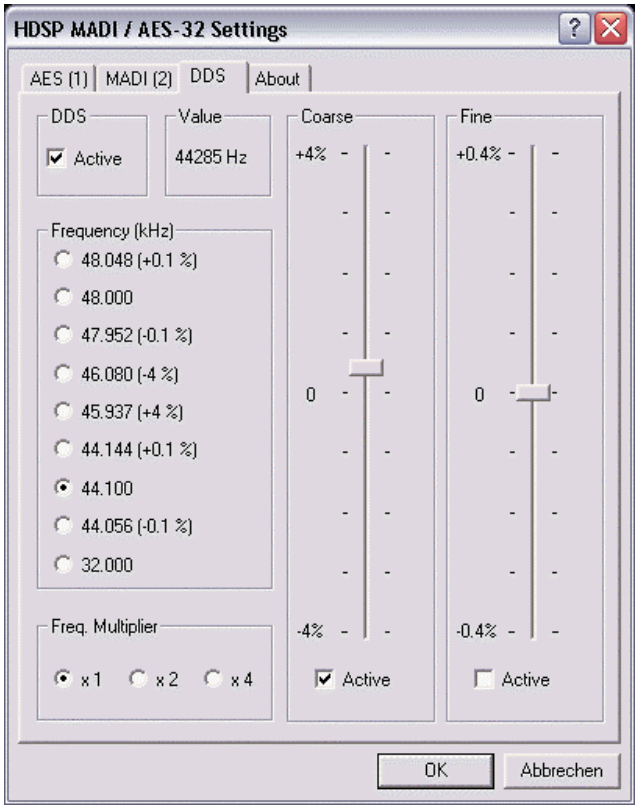
Value
Shows the sample rate as adjusted in this dialog. The sample rate is defined by the basic setting (frequency), the multiplier, and the position of the activated fader.

Frequency
Sets a fixed basic sample rate, which can be modified by multiplier and fader.

Freq. Multiplier
Changes the basic sample rate into Single, Double oder Quad Speed mode.

Coarse
Fader for coarse modification of the basic sample rate. Click *Active* to activate it. Minimum step size 1 Hz.

Fine
Fader for fine modification of the basic sample rate. Click *Active* to activate it. Minimum step size 1 Hz.



Notes on the faders
A mouse click within the fader area, above or below the fader know, will move the fader with the smallest step size up or down. Holding the Ctrl key while clicking will cause the fader to jump to its center (0) position.

11.3 Clock Modes - Synchronisation

AutoSync
The HDSP AES-32 has been equipped with *AutoSync*, an automatic clock source selection, which adopts the first available input with a valid digital signal as the clock reference input. The input currently used as sync reference is shown in the 'System Clock' status field, together with its current sample frequency.

AutoSync guarantees that normal record and record-while-play will always work correctly. In certain cases however, AutoSync may cause feedback in the digital carrier, so synchronization breaks down. To remedy this, switch the HDSP's clock mode over to 'Master'.

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'.

Thanks to its AutoSync technique and lightning fast PLLs, the HDSP is not only capable of handling standard frequencies, but also any sample rate between 28 and 200 kHz. Even the word clock input, which most users will use in varispeed operation, allows any frequency between 28 kHz and 200 kHz.

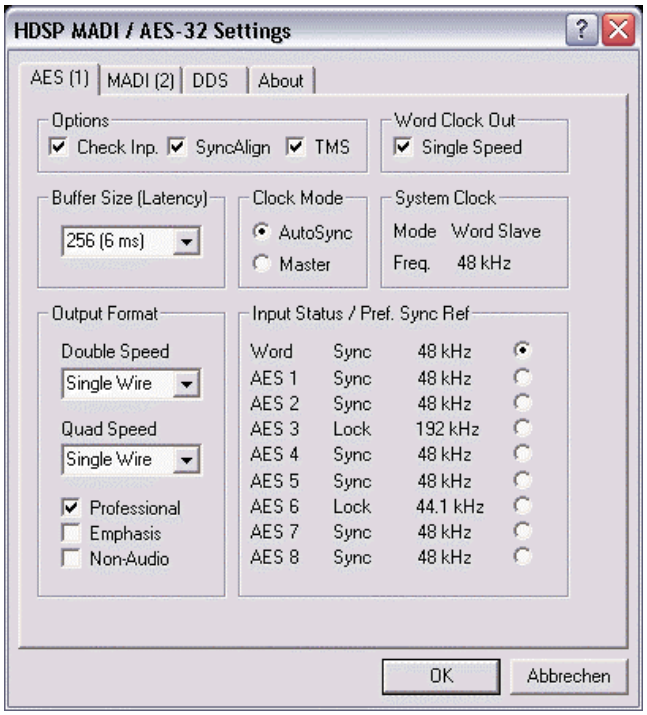
The HDSP AES-32's outstanding clock control allows for a synchronization of the output signal to the word clock's input signal not only at identical sample rates, but also at half, quarter, double and quad sample rates. A playback of 192 kHz can easily be synchronized via a 48 kHz word clock signal.

SyncCheck
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.

Remember that a digital system can only have one master! If the HDSP AES-32’s clock mode is set to Master, all other devices must be set to Slave.

RME’s exclusive **SyncCheck** technology (first implemented in the Hammerfall) enables an easy to use check and display of the current clock status. The SyncCheck field indicates whether no signal (No Lock), a valid signal (Lock) or a valid *and* synchronous signal (Sync) is present at each of the digital clock source inputs. The ‘AutoSync Ref’ display shows the current sync source and the measured frequency.

In practice, SyncCheck provides the user with an easy way of checking whether all digital devices connected to the system are properly configured. With SyncCheck, finally anyone can master this common source of error, previously one of the most complex issues in the digital studio world.



12. Operation and Usage

12.1 Playback

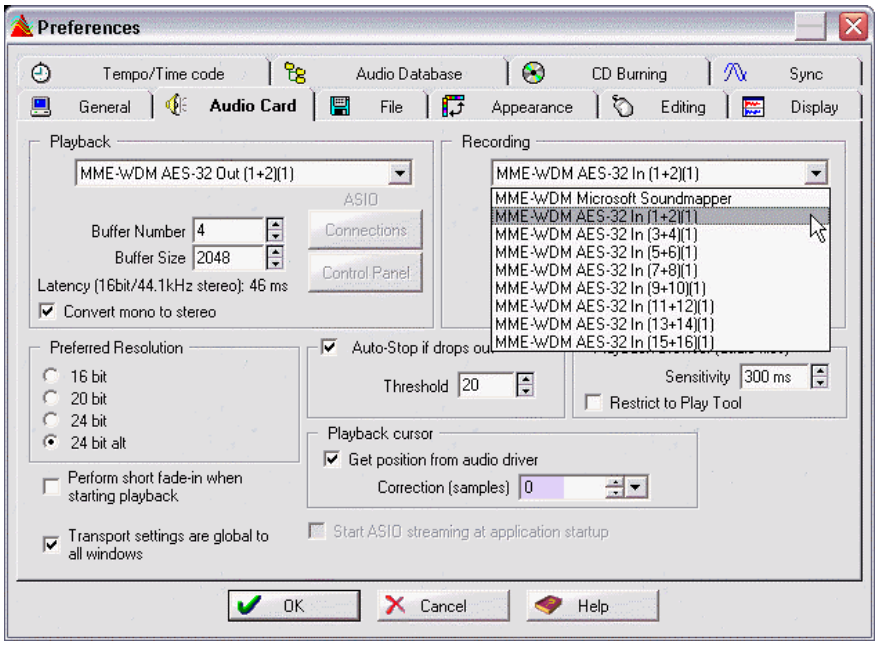
The HDSP system can play back audio data only in supported modes (channels, PCM) and formats (sample rate, bit resolution). Otherwise an error message appears (for example at 22 kHz and 8 bit).

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 strongly recommend switching off all system sounds (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 screenshot to the right shows a typical configuration dialog as displayed by a (stereo) wave editor. After selecting one of the eight stereo playback devices, audio data is sent to the according audio channels.

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'.



12.2 DVD-Playback (AC-3/DTS) 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 via the HDSP AES-32. For this to work an 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', 'Use SPDIF' 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 AES-32 automatically activates the non-audio bit in 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 AES-32. 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 AES-32.

The device selected as *Preferred Playback Device* defines the first playback channel. Choosing AES 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 - SL (Surround Left)
6 - SR (Surround Right)

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 AES-32. So when using AutoSync and/or word clock, the playback speed and pitch follows the incoming clock signal.

Note 3: PowerDVD 5 no longer supports a starting channel other than channel 1. Use TotalMix to send audio to any aother output port if necessary.

Note 4: In WinDVD 5 and 6, a channel-separated 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!**

12.3 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. Playback will be stuttering and audio will be distorted.

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.

Example: when you set the Hammerfall to 512 you can't use 128 in any program. But setting the buffer size 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 (like older VIA boards) suffer from crackling at settings below 512. Be sure to set the buffer size to 512 or higher in such a case (or get a new motherboard).

12.4 Multi-client Operation

RME audio interfaces 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!*

I.e. 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, this playback pair can't be used in Gigasampler/Studio (GSIF) nor under MME (WaveLab etc.) anymore. 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. Note that the inputs can be used at the same time, as the driver sends the data to all applications simultaneously.

ASIO-Multiclient
RME audio interfaces 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. Once again the same inputs can be used simultaneously.

RME's sophisticated tool *DIGICheck* is an exception to this rule. It operates like an ASIO host, using a special technique to access playback channels already occupied. Therefore DIGICheck is able to analyse and display playback data from any software, no matter which format the software uses.

12.5 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.

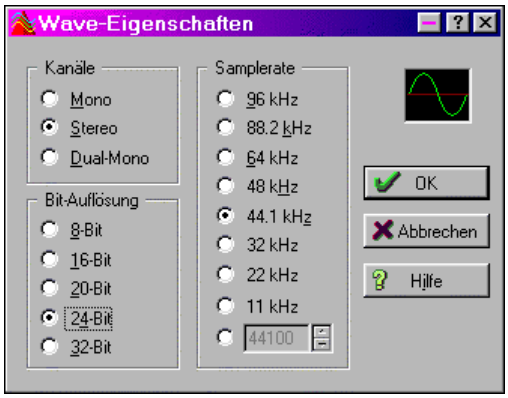
To take this into account, RME has included two unique features in the HDSP AES-32: a comprehensive I/O signal status display (showing sample frequency, lock and sync status) in the Settings dialog, and the protective *Check Input* function.

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 11, screenshot Settings) is useful as a quick display of the current configuration (the box itself and all connected external equipment). If no sample frequency is recognized, it will read ‘No Lock’.

This way, configuring any suitable audio application for digital recording is simple. After selecting the required input, HDSP AES-32 displays the current sample frequency. This parameter can then be changed in the application’s audio attributes (or similar) dialog.

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 24).

Currently two solutions exist which enable an *automated* control of real-time monitoring. ZLM 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 Magix), and can be activated using the global track option 'Hardware monitoring during Punch'. As ZLM is limited to MME, this mode is no longer supported since TotalMix version 2.3.

The second solution is Steinberg’s ASIO protocol with our ASIO 2.0 drivers and all ASIO 2.0 compatible programs (even Samplitude…). When 'ASIO Direct Monitoring' has been switched on, the input signal is routed in real-time to the output whenever a recording is started.

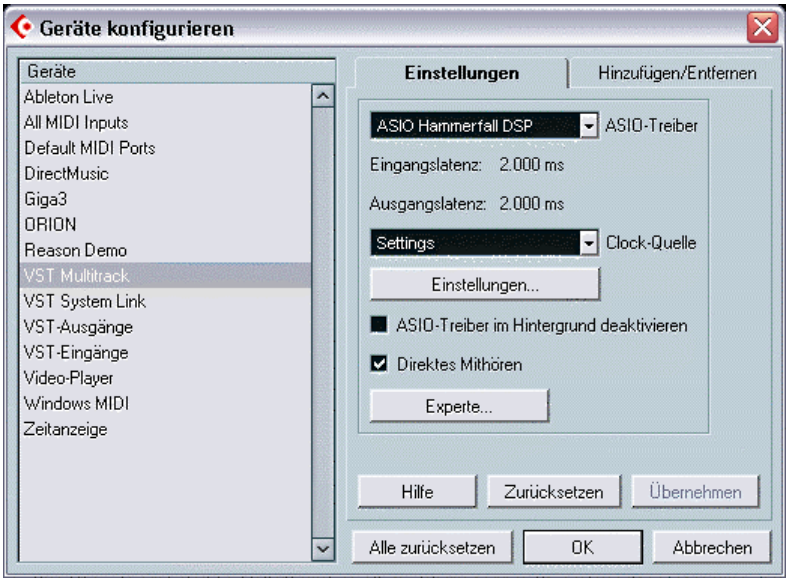
13. Operation under ASIO 2.0

13.1 General

Start the ASIO software and select **ASIO HDSP MADI/AES-32** as the audio I/O device. The 'ASIO system control' button opens the HDSP's Settings dialog (see chapter 11, Configuration).

Hammerfall DSP supports *ASIO Direct Monitoring* (ADM). Please note that currently Nuendo, Cubase and Logic either do not support ADM completely or error-free.

Using emulated MIDI drivers often causes a drift and delay between audio and MIDI. You should use non-emulated (MME) MIDI ports in such a case.



13.2 Known Problems

If a computer does not provide sufficient CPU-power and/or sufficient PCI-bus transfer rates, then drop outs, crackling and noise will appear. We recommend to deactivate all PlugIns to verify that these are not the reason for such effects.

Additional hard disk controllers, both on-board and PCI based, aften violate 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').

When using more than one HDSP system, all units have to be in sync, see chapter 15. Else a periodicly repeated noise will be heared.

Another common source of trouble is incorrect synchronization. ASIO does not support asynchronous operation, which means that the input and output signals not only have to use the same sample frequency, but also have to 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)

The GSIF interface of the HDSP AES-32 allows direct operation with Gigastudio, with up to 16 channels, 96 kHz and 24 bit. GSIF 2.0 is also supported with both audio and MIDI.

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.

If Gigastudio starts up properly, loads 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. Using multiple HDSP AES-32 / HDSP MADI

The current driver supports operation of up to three HDSP AES-32 in any combination with the HDSP MADI. Both HDSP AES-32 and HDSP MADI use the same driver, therefore can be used at the same time. Please note that only one TCO of one card can be used (of course). All units have to be in sync, i.e. have to receive valid sync information either via word clock or by using AutoSync and feeding synchronized signals.

- If one of the HDSP systems is set to clock mode Master, all others have to be set to clock mode AutoSync, and have to be synced from the master, for example by feeding word clock. The clock modes of all units have to be set up correctly in their Settings dialog.

- If all units are fed with a synchronous clock, i.e. all units show *Sync* in their Settings dialog, all channels can be used at once. This is especially easy to handle under ASIO, as the ASIO driver presents all units as one.

Note: TotalMix is part of the hardware of each HDSP system. Up to three mixers are available, but these are separated and can't interchange data. Therefore a global mixer for all units is not possible.

16. DIGICheck

The DIGICheck software is a unique utility developed for testing, measuring and analysing digital audio streams. Although this Windows software is fairly self-explanatory, it still includes a comprehensive online help. DIGICheck 4.42 operates as multi-client ASIO host, therefore can be used in parallel to any software, be it MME, ASIO or GSIF, with both inputs and outputs (!). The following is a short summary of the currently available functions:

- **Level Meter**. High precision 24-bit resolution, 2/8/64 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. Vertical and horizontal mode. Slow RMS and RLB weighting filter. Supports visualization according to the K-system.
- **Hardware Level Meter** for **Input**, **Playback** and **Output**. As above, received pre-calculated directly from the HDSP system hardware with near zero CPU load.
- **Spectral Analyser.** World wide unique 10-, 20- or 30-band display in analog bandpass-filter technology. 192 kHz-capable!
- **Vector Audio Scope**. World wide unique Goniometer showing the typical afterglow of an oscilloscope-tube. Includes Correlation meter and level meter.
- **Totalyser**. Spectral Analyser, Level Meter and Vector Audio Scope in a single window.
- **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.
- **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.

DIGICheck is conctantly improved. The latest version is always found on our website **www.rme-audio.de**, section **Downloads/Tools**.

17. Hotline – Troubleshooting

17.1 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 (for example in TotalMix).

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 HDSP system 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.

17.2 Installation

Hammerfall DSP is found in the Device Manager (>*Settings/ Control Panel/ System*<), category 'Sound-, Video- and Gamecontroller'. A double click on 'Hammerfall DSP AES-32' starts the properties dialog. Choosing 'Resources' shows Interrupt and Memory Range.

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

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

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

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

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

User's Guide

HDSP AES-32

Driver Installation and Operation – Mac OS X

18. Driver and Flash Update

18.1 Driver Installation

First fit the card (see 5. Hardware Installation), then switch on the computer and install the drivers from the RME Driver CD. The driver file is located in the folder **HDSP MADI AES32**. Installation works automatically by a double-click on the file **hdsp_madi_aes32.mpkg**.

RME recommends to download the latest driver version from the RME website! If done, the procedure is as follows:

Double-click onto **madi_aes_xx.gz** to expand the archive file to **madi_aes_xx.tar** and the folder **HDSP_MADI_xx**, which includes the driver file **hdsp_madi_aes32.mpkg**. Installation works automatically by a double-click on this file.

During driver installation the programs **Settings** and **Mixer** (TotalMix) will also be installed. Both programs start automatically as soon as a HDSP system is detected. They stay in the dock when exited, and remove themselves automatically from the dock when the HDSP system is removed.

Reboot the computer when installation is done.

18.2 Driver Update

In case of a driver update it's not necessary to remove the old driver first, it will be overwritten during the installation.

Exception: driver updates from version <2.5. Remove the former Settings dialog and TotalMix from the Login Items, and delete both files from your hard drive!

This driver version did not have the features AutoLoad, Dock Lock and AutoRemove. Therefore one has to make sure that both programs have been removed from the system, to prevent the old Settings dialog and TotalMix from being loaded.

18.3 Flash Update

The Flash Update Tool updates the HDSP AES-32 card to the latest firmware version. It requires an already installed driver.

Start the program **HDSP MADI AES-32 Flash**. The Flash Update Tool displays the current revision of the HDSP AES-32 interface, and whether it needs an update or not. If so, then 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 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.

19. Configuring the HDSP AES-32

19.1 Settings Dialog

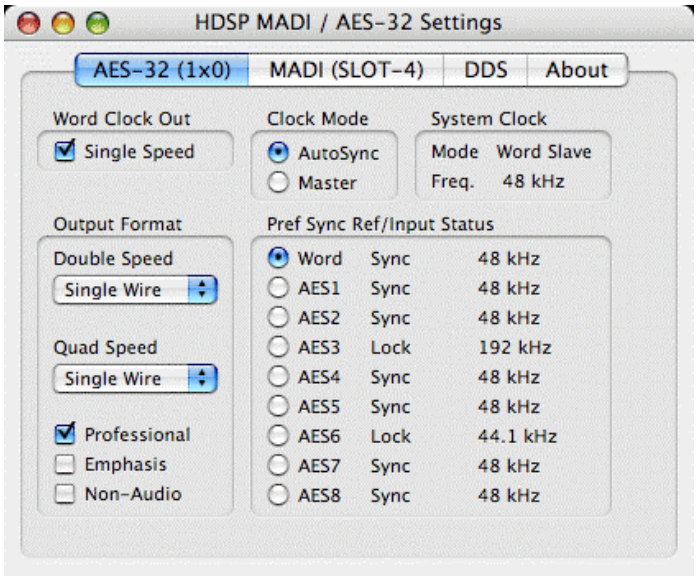
Configuring the HDSP AES-32 is done via its own settings dialog. The panel 'Settings' can be opened by clicking on the hammer icon in the dock. The mixer of the HDSP AES-32, TotalMix, can be opened by clicking on the mixer icon in the dock.

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
- Configuration of digital I/Os
- Synchronization behaviour
- State of input and output
- Current sample rate

Any changes performed in the Settings dialog are applied immediately - confirmation (e.g. by 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, and the status of all digital signals.



Quick Boot
All the card's settings described below are stored in a hardware memory, and are loaded immediately after a power-on of the computer. In clock mode Master even the last used sample rate is set. Directly after switching on the computer, a stable and predictable clock state is found at the HDSP AES-32's outputs. This advanced technology completely eliminates disturbing noises and clock network problems during power-up or re-boot.

Word Clock Out
The word clock output signal usually equals the current sample rate. Selecting *Single Speed* causes the output signal to always stay within the range of 32 kHz to 48 kHz. So at 96 kHz and 192 kHz sample rate, the output word clock is 48 kHz.

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

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

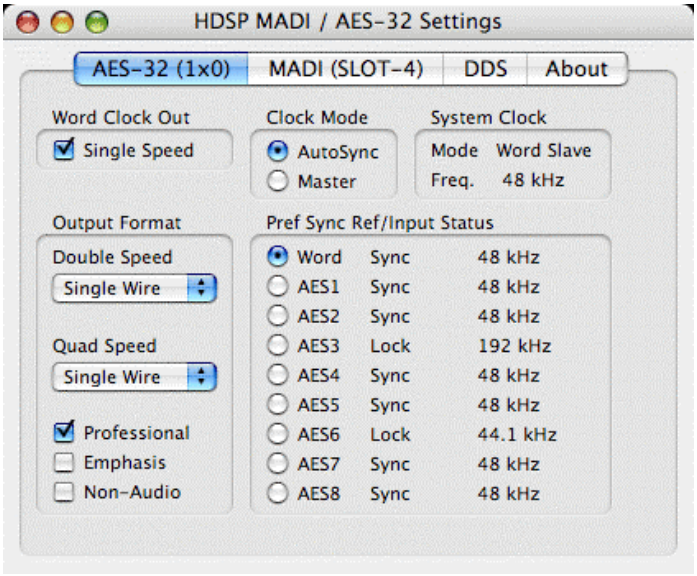
Output Format
Double Speed: Sample rates in the range of 64 kHz to 96 kHz can also be transmitted using the standard 48K frame. Note that when selecting *Double Wire* the number of output channels is reduced to 8.

Quad Speed: Sample rates in the range of 128 kHz to 192 kHz can also be transmitted using either the standard 48K frame, or the 96K frame. Note that when selecting *Double Wire* the number of output channels is reduced to 8. When selecting *Quad Wire* the number of remaining output channels is 4.

For further details about the settings ‘Professional’, ‘Emphasis’ and ‘Non-Audio’, please refer to chapter 22.2.

Input Status
SyncCheck indicates whether there is a valid signal (Lock) for the inputs Word Clock and AES 1 to 8, or if there is a valid *and* synchronous signal (Sync). The *System Clock Mode* display shows the input and frequency of the current sync source. Each input has its own frequency measurement and display of the input signal's current sample rate.

Pref. Sync Ref.
Used to pre-select the desired clock source. If the selected source isn't available, the unit will change to the next available one. The current clock source and sample rate is displayed in the *System Clock* field.



19.2 Settings dialog - DDS

Usually soundcards and audio interfaces generate their internal clock (master mode) by a quartz. Therefore the internal clock can be set to 44.1 kHz or 48 kHz, but not to a value in between. SteadyClock, RME's sensational Low Jitter Clock System, is based on a *Direct Digital Synthesizer* (DDS). This superior circuitry can generate nearly any frequency with highest precision.

DDS has been implemented into the HDSP AES-32 with regard to the needs of professional video applications, as well as to maximum flexibility. The dialog DDS includes both a list of typical video frequencies (so called pull up/pull down at 0.1% and 4%) and two faders, which allow to freely change the basic sample rate in steps of 1 Hz (!).

The DDS dialog requires the HDSP AES-32 to be in clock mode Master! The frequency setting will only be applied to this one specific card!

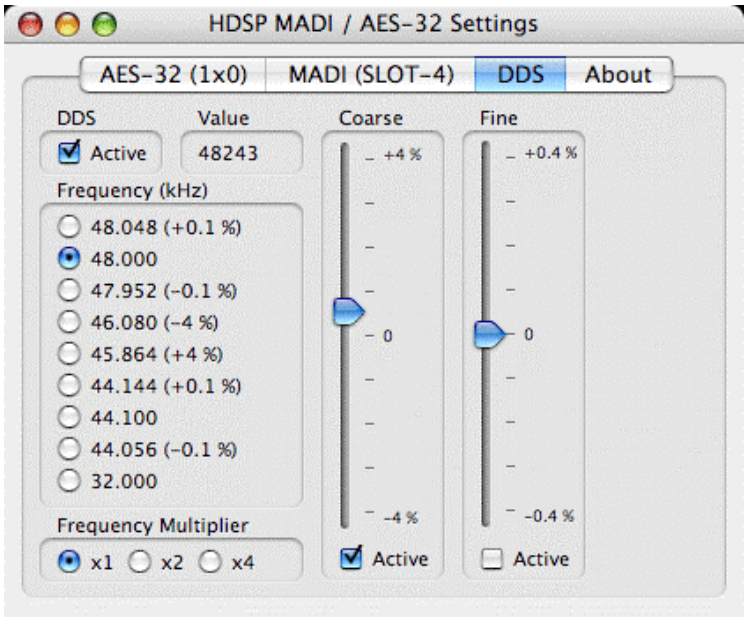
Changing the sample rate in bigger steps during record/playback often results in a loss of audio, or brings up warning messages of the audio software. Therefore the desired sample rate should be set at least coarsely before starting the software.

DDS
Activates all settings of this dialog.

Value
Shows the sample rate as adjusted in this dialog. The sample rate is defined by the basic setting (Frequency), the multiplier, and the position of the activated fader.

Frequency
Sets a fixed basic sample rate, which can be modified by multiplier and fader.

Freq. Multiplier
Changes the basic sample rate into Single, Double oder Quad Speed mode.



Coarse
Fader for coarse modification of the basic sample rate. Click *Active* to activate it. Minimum step size 1 Hz.

Fine
Fader for fine modification of the basic sample rate. Click *Active* to activate it. Minimum step size 1 Hz.

Notes on the faders
A mouse click within the fader area, above or below the fader know, will move the fader with the smallest step size up or down. Holding the Ctrl key while clicking will cause the fader to jump to its center (0).

Application examples
DDS allows for a simultaneous change of speed and tune during record and playback. From alignment to other sources up to creative effects – everything is possible..

DDS allows to intentionally de-tune the complete DAW. This way, the DAW can match instruments which have a wrong or unchangeable tuning.

DDS allows to define a specific sample rate. This feature can be is useful in case the system randomly changes the sample rate – for unknown reasons. It also prevents a change from Double Speed (96 kHz) to Single Speed (48 kHz), which would cause configuration and routing problems by the changed amount of AES channels when using Double and Quad Wire techniques.

19.3 Clock Modes - Synchronisation

AutoSync
The HDSP AES-32 has been equipped with *AutoSync*, an automatic clock source selection, which adopts the first available input with a valid digital signal as the clock reference input. The input currently used as sync reference is shown in the *AutoSync Ref* status field, together with its current sample frequency.

AutoSync guarantees that normal record and record-while-play will always work correctly. In certain cases however, AutoSync may cause feedback in the digital carrier, so synchronization breaks down. To remedy this, switch the HDSP's clock mode over to 'Master'.

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'.

Thanks to its AutoSync technique and lightning fast PLLs, the HDSP is not only capable of handling standard frequencies, but also any sample rate between 28 and 200 kHz. Even the word clock input, which most users will use in varispeed operation, allows any frequency between 28 kHz and 200 kHz.

The HDSP AES-32's outstanding clock control allows for a synchronization of the output signal to the word clock's input signal not only at identical sample rates, but also at half, quarter, double and quad sample rates. A playback of 192 kHz can easily be synchronized via a 48 kHz word clock signal.

SyncCheck
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.

Remember that a digital system can only have one master! If the HDSP AES-32's clock mode is set to Master, all other devices must be set to Slave.

RME's exclusive **SyncCheck** technology (first implemented in the Hammerfall) enables an easy to use check and display of the current clock status. The SyncCheck field indicates whether no signal (No Lock), a valid signal (Lock) or a valid *and* synchronous signal (Sync) is present at each of the digital clock source inputs. The 'AutoSync Ref' display shows the current sync source and the measured frequency.

In practice, SyncCheck provides the user with an easy way of checking whether all digital devices connected to the system are properly configured. With SyncCheck, finally anyone can master this common source of error, previously one of the most complex issues in the digital studio world.

20. Mac OS X FAQ

20.1 Round about Driver Installation

The driver with the file suffix **gz** provided by RME is a compressed **TAR** archive. TAR bundles multiple files and folders into one file, but does not save memory space nor download time. Both TAR and gz are supported natively by OS X, a double click on the file is all you need to do.

Older browsers do not recognize **gz** as an archive, loading the file as a document. This results in a cryptic looking text within the browser window. Downloading the file can be done via the right mouse key, **Save Target as**. Despite this procedure, some older browsers like Netscape 4.78 will not save the file correctly - the archive will be corrupted.

The driver consists of a package file (pkg), which contains various folders and files, similar to TAR. A double click will start the OS X installer. To save you the hassle of installing both audio and MIDI drivers separately, the HDSP driver contains an additional **meta package** (mpkg), that points to the single packages. Those single packages are not shown in the Finder, as they reside within the invisible folder '.contained_packages'. Only the mpkg is visible. Important: an installation can only be done with the **complete** folder. If only the mpkg is copied to a different place, it will not find the single driver packages!

The actual audio driver appears as a kernel extension file. The installer copies it to **>System/ Library/ Extensions<**. Its name is **HDSPMADI.kext**. It is visible in the Finder, allowing you to verify date and driver version. Yet, in fact this again is a folder containing subdirectories and files.

Nonetheless, this 'driver file' can be removed by simply dragging it to the trash bin. This can be helpful in case a driver installation fails. An incomplete installation can currently (10.3.2) only be detected indirectly: The installation routine does not open a message window with a note about a restart of the computer. This indicates that the driver file was not copied and the driver was not installed!

Several users have observed that the installation routine occasionally stops and no longer works correctly. This can be fixed by removing the corresponding extension file prior to installation. In some cases, also (or only) a repair of the **disk permission** will help.

We have also received reports saying the driver update could not be installed on the system disk - shown red crossed during the installation. Repairing permission may also help here. If not, we're sorry, but have to recommend to contact Apple. Our driver has no knowledge of folders, disks etc., the installation is handled completely by the OS X installer.

20.2 MIDI doesn't work

In some cases MIDI does not work after the installation of the HDSP driver. To be precise, applications do not show an installed MIDI port. The reason for this is usually visible within the **Audio MIDI Setup**. It displays no RME MIDI device, or the device is greyed out and therefore inactive. Mostly, removing the greyed out device and searching for MIDI devices again will solve the problem. If this does not help, we recommend manual removal of the MIDI driver and reinstallation of the complete driver. Otherwise repairing permissions may help.

The HDSP MIDI driver is a plugin. During installation it will be copied to **>Library/ Audio/ MIDI Drivers<**. It's name is **Hammerfall DSP MIDI.plugin**. The file can be displayed in the Finder and also be removed by simply dragging it to the trash bin.

20.3 Supported Sample Rates

RME's Mac OS X driver supports all sampling frequencies provided by the hardware. Besides **192 kHz** and **96 kHz** this also includes **32 kHz** and **64 kHz**.

But not every software will support all the hardware's sample rates. For example Spark does not display 32 kHz and 64 kHz. The hardware's capabilities can easily be verified in the **Audio MIDI Setup**. Select **Audio devices** under **Properties of:** and choose the Hammerfall DSP. A click on **Format** will list the supported sample frequencies.

If the unit is in clock mode **Master**, selecting a sample rate will immediately set the device to this frequency, which can be verified in the HDSP's settings dialog (System Clock). **Format** thus allows you to activate any sampling frequency quickly and easily.

20.4 Repairing Disk Permissions

Repairing permission can solve problems with the installation process - plus many others. To do this, launch **Disk Utility** located in **Utilities**. Select your system drive in the drive/volume list to the left. The **First Aid** tab to the right now allows you to check and repair disk permissions.

20.5 PCI card and PCI slot compatibility

Unfortunately not every RME card will work in every PCI slot of an Apple computer. To our knowledge, the current Hammerfall DSP systems can be used in any PCI slot of all G4 and G5 models. In case additional PCI cards of any manufacturer are used, it might happen that the RME card is no longer found by the system. Swapping cards between slots can be helpful in this case.

20.6 Various Information

The driver requires 10.2.8 or higher. Older versions of OS X are not and will not be supported.

Via **>System Preferences/ Audio-MIDI Setup<** the hardware can be configured for the system wide usage. Programs that don't support card or channel selection will use the device selected as **Standard-Input** and **Standard-Output**. (Soundstudio, Mplayer, Amplitube etc.).

In the lower part of the window, the audio hardware's capabilities are shown and can be changed in some cases. On the record side no changes are possible. Programs that don't support channel selection will always use channels 1/2, the first stereo pair.

Since OS X 10.3 playback can be configured freely and to any of the available playback channels. This is done via **Speaker Setup**. Even multichannel playback (Surround, DVD Player) can be set up easily.

OS X supports more than one audio device. Their simultaneous usage within one program had been limited to Motu's Digital Performer until 10.3.9. Since 10.4 (Tiger) Core Audio offers the function **Aggregate Devices**, which allows to combine several devices into one, so that a multi-device operation is now possible with any software.

The Hammerfall DSP driver adds a number to each unit, so they are fully accessible in any multicard-capable software.

Our experience so far is that using more than one HDSP AES-32 at higher sample rates will work only on dedicated server systems having multiple PCI busses. A usage of two cards at full track count at 192 kHz is therefore difficult to achieve on current Mac computers. See chapter 30.7, *PCI – Performance*.

21. Hotline – Troubleshooting

The newest information can always be found on our website www.rme-audio.com, section Support, Macintosh OS.

Playback works, but record doesn't:

- Check that there is a valid signal at the input.
- 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 application.
- Try different cables to rule out any defects here.
- Use an external FireWire drive for the audio data. Internal SATA drives overload the system bus in some Macs, thus disturb PCI audio.

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

- Is Hammerfall DSP listed in the System Profiler/PCI? (Vendor 10EE, Device ID 3FC6).
- Has Hammerfall DSP been selected as current playback device in the audio application?

User's Guide

HDSP AES-32

Connections and TotalMix

22. Connections

22.1 AES/EBU Inputs

The AES/EBU inputs are provided via 25 pin D-sub connectors with Tascam pinout (also used by Digidesign). A digital breakout cable will provide 4 female (and 4 male) XLR connectors per D-sub connector. Every input is transformer-balanced and ground-free. Channel status and copy protection are being ignored.

The inputs can be used in any combination, e. g. it is sufficient to connect an input signal only to input 3. In slave mode, this input is automatically being used as clock source. If more than one signal is present, the one furthest left is being used as clock source, i. e. the active input with the lowest number.

The HDSP AES-32 supports all currently known formats in the range of 32 kHz up to 192 kHz, including sample multiplexing:

- Single Wire: 16 channels 32 kHz – 192 kHz. 2 channels per AES wire. The effective sample frequency equals the clock on the AES wire.

- Double Wire: 8 channels 64 kHz – 192 kHz. 1 channel per AES wire. The effective sample frequency is double the clock of the AES wire.

- Quad Wire: 4 channels 128 kHz – 192 kHz. 1 channel via 2 AES wires. The effective sample frequency is four times the clock of the AES wire.

Rearranging the formats Double and Quad Wire to Single Wire is lossless, the existing samples are just re-ordered again. Information on the distribution of the samples in Double and Quad Wire mode is found in chapter 22.2, AES/EBU Outputs.

The selection of the corresponding format is fully automated by the relation between input signal sample rate and the sample rate requsted by the application. Example: If a 48 kHz AES signal is detected at the input, and the audio software is configured to use a sample rate of 96 kHz, the hardware expects the input signal to be in Double Wire format. A ratio of 48 kHz to 192 kHz will activate Quad Wire mode. The hardware processes the sample multiplexing of the input data. The number of input channels shown in TotalMix will be reduced to 8 or 4, so the currently active mode is easy to recognize.

Emphasis
AES/EBU and SPDIF can contain an Emphasis information. Audio signals with Emphasis have a strong high frequency boost and thus require a high frequency attenuation on playback.

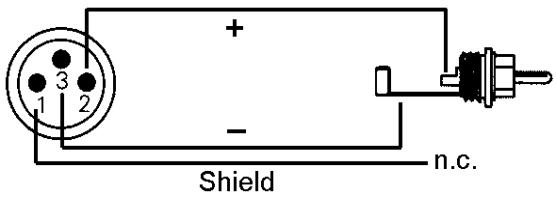
An Emphasis indication on an input is lost as there exists no standardized interface on comuters to pass on this information!

Pinout of the D-sub connector, Inputs

Signal	In 1/2+	In 1/2-	In 3/4+	In 3/4-	In 5/6+	In 5/6-	In 7/8+	In 7/8-
D-Sub	24	12	10	23	21	9	7	20

GND is connected to pins 2, 5, 8, 11, 16, 19, 22, 25. Pin 13 is not connected.

Thanks to a highly sensitive input stage, also SPDIF signals can be processed by using a simple cable adapter phono/XLR. To achieve this, pins 2 and 3 of a male 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.



22.2 AES/EBU Outputs

The AES/EBU outputs are provided via 25 pin D-sub connectors with Tascam pinout (also used by Digidesign). A digital breakout cable will provide 4 male (and 4 female) XLR connectors per D-sub connector. Each output is transformer-balanced, ground-free and compatible to all devices with AES/EBU ports.

If *Output Format Professional* is chosen, the output level is almost 5 Volt. If deselected, the output signal will have a channel status compatible to SPDIF. As far as we know, every SPDIF device should be capable of handling an input signal of up to 5 Volt instead of the usual 0.5 Volt. Nevertheless the output level will be reduced to 2 Volt in this case.

Besides the audio data, digital signals in SPDIF or AES/EBU format contain a channel status coding, which is being used for transmitting further information. The output signal coding of the HDSP AES-32 has been implemented according to AES3-1992 Amendment 4:

- 32 kHz, 44.1 kHz, 48 kHz, 64 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz according to the current sample rate
- Audio use
- No Copyright, Copy permitted
- Format Consumer or Professional
- Category General, Generation not indicated
- 2-Channel, No Emphasis oder 50/15 µs
- Single Channel Double Sampling Frequency Mode (Double Wire)
- Aux Bits Audio use, 24 Bit
- Origin: HDSP

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

The status 'Professional' should always be active when sending data to a device with AES/EBU inputs (when the XLR connectors are used).

Pinout of the D-sub connector, Outputs

Signal	Out 1/2+	Out 1/2-	Out 3/4+	Out 3/4-	Out 5/6+	Out 5/6-	Out 7/8+	Out 7/8-
D-Sub	18	6	4	17	15	3	1	14

GND is connected to pins 2, 5, 8, 11, 16, 19, 22, 25. Pin 13 is not connected.

Connecting devices with coaxial SPDIF ports to the HDSP AES-32 outputs is accomplished by a simple cable adapter XLR/phono, as described above. Just use a female XLR connector.

The outputs of the HDSP AES-32 support all currently known formats in the range of 32 kHz up to 192 kHz. The current output format is defined in the section *Output Format* depending on the sample rate:

Double Speed
When the card operates in the range of 64 kHz to 96 kHz, Single Wire and Double Wire are available as output format.

- Single Wire: 16 channels. 2 channels per AES wire. The effective sample frequency equals the clock on the AES wire (64 kHz – 96 kHz).

- Double Wire: 8 channels. 1 channel per AES wire. The effective sample frequency is double the clock of the AES wire (32 kHz – 48 kHz).

Quad Speed
When the card operates in the range of 128 kHz to 192 kHz, Single Wire, Double Wire and Quad Wire are available as output format.

- Single Wire: 16 channels. 2 channels per AES wire. The effective sample frequency equals the clock on the AES wire (128 kHz – 192 kHz).

- Double Wire: 8 channels. 1 channel per AES wire. The effective sample frequency is double the clock of the AES wire (64 kHz – 96 kHz).

- Quad Wire: 4 channels. 1 channel via 2 AES wires. The effective sample frequency is four times the clock of the AES wire (32 kHz – 48 kHz).

All these conversions are lossless. Only the existing samples are spread on all the channels. The default setting is Single Wire. The card then operates gapless in the range of 32 kHz up to 200 kHz with full 16 channels.

The distribution of the samples performed in Double Wire mode equals the S/MUX method and is done as follows:

Channel	**L**	**R**	**L**	**R**	**L**	**R**	**L**	**R**
Port	**1**	**1**	**2**	**2**	**3**	**3**	**4**	**4**
Samples	**1a**	**1b**	**2a**	**2b**	**3a**	**3b**	**4a**	**4b**

The distribution of the samples performed in the Quad Wire mode equals the S/MUX4 method and is done as follows:

Channel	**L**	**R**	**L**	**R**	**L**	**R**	**L**	**R**
Port	**1**	**1**	**2**	**2**	**3**	**3**	**4**	**4**
Samples	**1a**	**1b**	**1c**	**1d**	**2a**	**2b**	**2c**	**2d**

22.3 MIDI

The HDSP AES-32 offers two MIDI I/O via 5-pin DIN connectors. The MIDI ports are added to the system by the driver. Using MIDI capable software, these ports can be accessed under the name *HDSP AES-32 MIDI*. Using more than one HDSP AES-32, a consecutive number is added to the port name, like *AES-32 MIDI In 1 (2)* etc.

The MIDI In port is available for both GSIF (GSIF-2 Low Latency) and standard MME MIDI simultaneously.

23. Word Clock

23.1 Word Clock Input and Output

SteadyClock guarantees an excellent performance in all clock modes. Based on the highly efficient jitter suppression, the HDSP AES-32 refreshes and cleans up any clock signal, and provides it as reference clock at the BNC output (see chapter 30.6).

Input
The HDSP AES-32's transformer isolated word clock input is active when *Pref. Sync Ref* in the Settings dialog has been switched to *Word Clock*, the clock mode *AutoSync* has been activated, and a valid word clock signal is present. The signal at the BNC input can be Single, Double or Quad Speed, the HDSP AES-32 automatically adapts to it. As soon as a valid signal is detected, the Settings dialog shows either Lock or Sync (see chapter 30.2).

Thanks to RME's *Signal Adaptation Circuit*, the word clock input still works correctly even with heavily mis-shaped, dc-prone, too small or overshoot-prone signals. Thanks to automatic signal centering, 300 mV (0.3V) input level are sufficient in principle. An additional hysteresis reduces sensitivity to 1.0 V, so that over- and undershoots and high frequency disturbances don't cause a wrong trigger.

The word clock input is shipped as high impedance type (not terminated). An internal jumper besides the BNC socket allows to activate proper termination (75 Ohm).

Output
The word clock output of the HDSP AES-32 is constantly active, providing the current sample frequency as word clock signal. As a result, in Master mode the provided word clock is defined by the currently used software or the DDS dialog. In Slave mode the provided frequency is identical to the one present at the currently chosen clock input. When the current clock signal fails, the HDSP AES-32 switches to Master mode and adjusts itself to the next, best matching frequency (44.1 kHz, 48 kHz etc.).

Selecting the options Double Wire or Quad Wire in the Settings dialog the output frequency is changed to always be the same as the sample rate on the AES outputs. At 192 kHz operation with activated *Double Wire* operation the AES-32 will generate a word clock signal of 96 kHz.

The received word clock signal can be distributed to other devices by using the word clock output. With this the usual T-adapter can be avoided, and the HDSP AES-32 operates as *Signal Refresher*. This kind of operation is highly recommended, because

- input and output are phase-locked and in phase (0°) to each other
- SteadyClock removes nearly all jitter from the input signal
- the exceptional input (1 Vpp sensitivity instead of the usual 2.5 Vpp, dc cut, Signal Adaptation Circuit) plus SteadyClock guarantee a secure function even with highly critical word clock signals

Thanks to a low impedance, but short circuit proof output, the HDSP AES-32 delivers 4 Vpp to 75 Ohms. For wrong termination with 2 x 75 Ohms (37.5 Ohms), there are still 3.3 Vpp fed into the network.

23.2 Technical Description and Usage

In the analog domain one can connect any device to another device, a synchronization is not necessary. Digital audio is different. It uses a clock, the sample frequency. The signal can only be processed and transmitted when all participating devices share the same clock. If not, the signal will suffer from wrong samples, distortion, crackle sounds and drop outs.

AES/EBU, SPDIF, ADAT and MADI are self-clocking, an additional word clock connection in principle isn't necessary. But when using more than one device simultaneously problems are likely to happen. For example any self-clocking will not work in a loop cabling, when there is no 'master' (main clock) inside the loop. Additionally the clock of all participating devices has to be synchronous. This is often impossible with devices limited to playback, for example CD players, as these have no SPDIF input, thus can't use the self clocking technique as clock reference.

In a digital studio synchronisation is maintained by connecting all devices to a central sync source. For example the mixing desk works as master and sends a reference signal, the word clock, to all other devices. Of course this will only work as long as all other devices are equipped with a word clock or sync input, thus being able to work as slave (some professional CD players indeed have a word clock input). Then all devices get the same clock and will work in every possible combination with each other.

Remember that a digital system can only have one master!

But word clock is not only the 'great problem solver', it also has some disadvantages. The word clock is based on a fraction of the really 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 multiple times higher than the one of 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 AES-32. Combining the advantages of modern and fastest digital technology with analog filter techniques, re-gaining a low jitter clock signal of 22 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.

23.3 Cabling 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.

Ideally, the word clock signal is a 5 Volt square wave with the frequency of the sample rate, of which the harmonics go up to far above 500 kHz. To avoid voltage loss and reflections, both the cable itself and the terminating resistor at the end of the chain 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.

Unfortunately there are still many devices on the market, even newer digital mixing consoles, which are supplied with a word clock output that can only be called unsatisfactory. If the output breaks down to 3 Volts when terminating with 75 Ohms, you have to take into account that a device, of which the input only works from 2.8 Volts and above, does not function correctly already after 3 meter cable length. So it is not astonishing that because of the higher voltage, word clock networks are in some cases more stable and reliable if cables are not terminated at all.

Ideally all outputs of word clock delivering devices are designed with very low impedance, but all word clock inputs with high impedance, in order to not weaken the signal on the chain. But there are also negative examples, when the 75 Ohms are built into the device and cannot be switched off. In this case the network load is often 2 x 75 Ohms, and the user is forced to buy a special word clock distributor. Note that such a device is generally recommended for bigger studios.

The HDSP AES-32's word clock input can be high-impedance or terminated internally, ensuring maximum flexibility. If termination is necessary (e.g. because the card is the last device in the chain), activate the switch TERM between the BNC jacks on the Expansion Board so that the yellow TERM LED lights up (see chapter 23.1).

In case the HDSP AES-32 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). Of course devices with internal termination do not need T-adaptor and terminator plug.

Due to the outstanding SteadyClock technology of the HDSP AES-32, we recommend not to pass the input signal via T-adapter, but to use the card's word clock output instead. Thanks to SteadyClock, the input signal will both be freed from jitter and - in case of loss or drop out – be reset to a valid frequency.

23.4 Operation

The HDSP AES-32's word clock input is active when *Pref. Sync Ref* in the Settings dialog has been set to *Word Clock*, the clock mode *AutoSync* has been activated, and a valid word clock signal is present. The signal at the BNC input can be Single, Double or Quad Speed, the HDSP automatically adapts to it. As soon as a valid signal is detected, the Settings dialog shows either Lock or Sync (see chapter 30.2). In the status display *System Clock* the display changes to *Word*. This message allows the user to check whether a valid word clock signal is present and is currently being used.

In *Input Status* the frequency of the reference signal is measured by the hardware. In this case the frequency of the current word clock signal is measured and displayed.

24. TotalMix: Routing and Monitoring

24.1 Overview

The HDSP AES-32 includes a powerful digital real-time mixer, the *Hammerfall DSP mixer,* based on RME's unique, sample-rate independent **TotalMix** technology. It allows for practically unlimited mixing and routing operations, with all inputs and playback channels simultaneously, to any hardware outputs.

Here are some typical applications for TotalMix:

- Setting up delay-free submixes (headphone mixes). The HDSP AES-32 allows for up to 8 fully independent stereo submixes. On an analog mixing desk, this would equal 16 (!) Aux sends.

- Unlimited routing of inputs and outputs (free utilisation, patchbay functionality).

- Distributing signals to several outputs at a time. TotalMix offers state-of-the-art splitter and distributor functions.

- Simultaneous playback of different programs using only one stereo output. The ASIO multi-client driver allows to use several programs at the same time, but only on different playback channels. TotalMix provides the means to mix and monitor these on a single stereo output.

- Mixing of the input signal to the playback signal (complete ASIO Direct Monitoring). RME not only is *the* pioneer of ADM, but also offers the most complete implementation of the ADM functions.

- Integration of external devices. Use TotalMix to insert external effects devices, be it in the playback or in the record path. Depending on the current application, the functionality equals insert or effects send and effects return, for example as used during real-time monitoring when adding some reverb to the vocals.

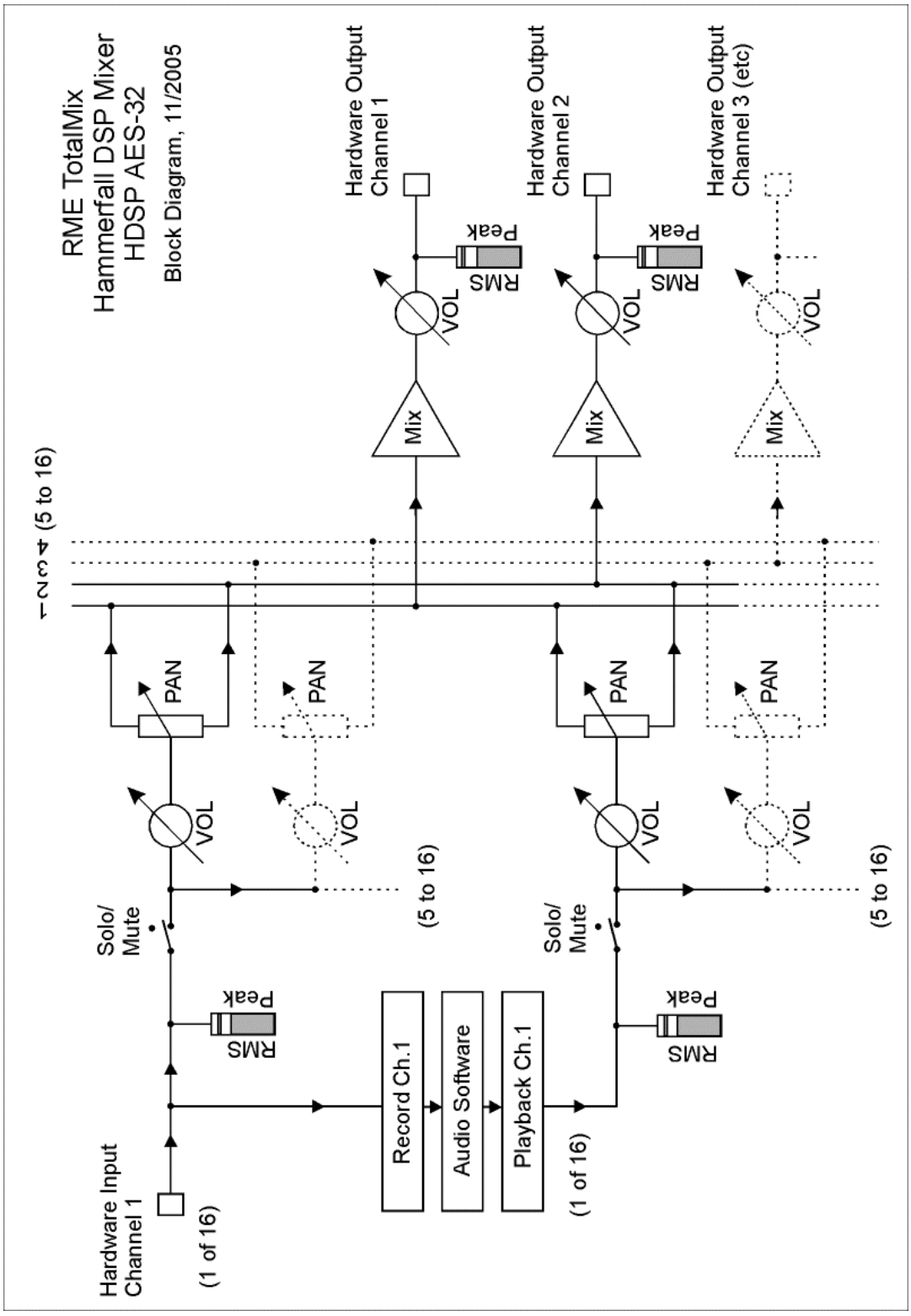
Every single input channel, playback channel and hardware output features a Peak and RMS level meter, calculated in hardware (hardware output is Peak only). These level displays are very useful to determine the presence and routing destinations of the audio signals.

For a better understanding of the TotalMix mixer you should know the following:

- As shown in the block diagram (next page), the record signal usually stays un-altered. TotalMix does not reside within the record path, and does not change the record level or the audio data to be recorded (exception: loopback mode).

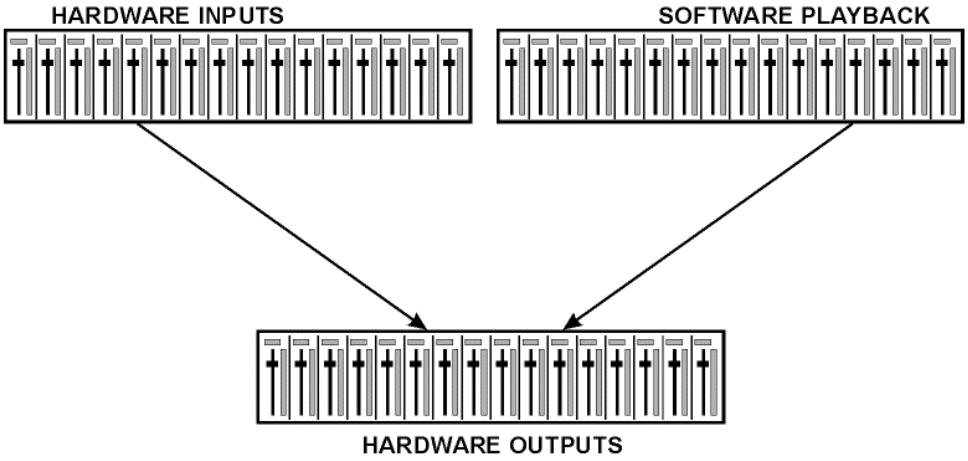
- The hardware input signal can be passed on as often as desired, even with different levels. This is a big difference to conventional mixing desks, where the channel fader always controls the level for all routing destinations simultaneously.

- The level meter of inputs and playback channels are connected pre-fader, to be able to visually monitor where a signal is currently present. The level meters of the hardware's outputs are connected post-fader, thus displaying the actual output level.

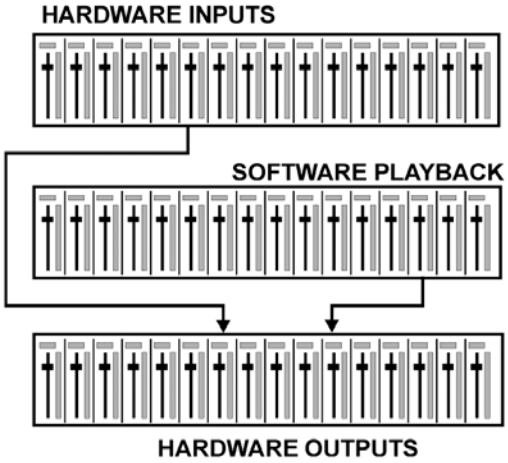

RME TotalMix
Hammerfall DSP Mixer
HDSP AES-32
Block Diagram, 11/2005
Hardware Output
Channel 1
Hardware Output
Channel 2
Hardware Output
Channel 3 (etc)
Peak
RMS
VOL
Mix
PAN
(5 to 16)
Solo/
Mute
Hardware Input
Channel 1
(1 of 16)
Record Ch.1
Audio Software
Playback Ch.1

24.2 The User Interface

The visual design of the TotalMix mixer is a result of its capability to route hardware inputs and software playback channels to any hardware output. The HDSP AES-32 provides 16 input channels, 16 software playback channels, and 16 hardware output channels:



Although 32 channels will fit on some screens side by side, such an arrangement provides no useful overview. The input channels should be placed above the corresponding output channel. Therefore, the channels have been arranged as known from an *Inline* desk, so that the row *Software Playback* equals the *Tape Return* of a real mixing desk:



- Top row: Hardware inputs. The level shown is that of the input signal, i. e. fader independent. Via fader and routing field, any input channel can be routed and mixed to any hardware output (bottom row).

- Middle row: Playback channels (playback tracks of the audio software). Via fader and routing field, any playback channel can be routed and mixed to any hardware output (third row).

- Bottom row (third row): Hardware outputs. Here, the total level of the output can be adjusted. This may be the level of connected loudspeakers, or the necessity to reduce the level of an overloaded submix.

The following chapters explain step by step all functions of the user interface.

24.3 Elements of a Channel

A single channel consists of various elements:

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

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

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

Next is the fader with a level meter. The meter shows both peak values (zero attack, 1 sample is enough for a full scale display) 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. Selecting one or more channels is done by clicking on the white label which turns orange then. A click in the third row with pressed Ctrl-key activates internal loopback mode, the label turns red. A right mouse click opens a dialog to type in a new name.

The black area (routing field) shows the current routing target. A mouse click opens the routing window to select a routing target. The list shows all currently activated routings by checkmarks in front of the routing targets.

24.4 Tour de TotalMix

This chapter is a practical guide and introduction on how to use TotalMix, and on how TotalMix works.

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, and activating phones monitoring.

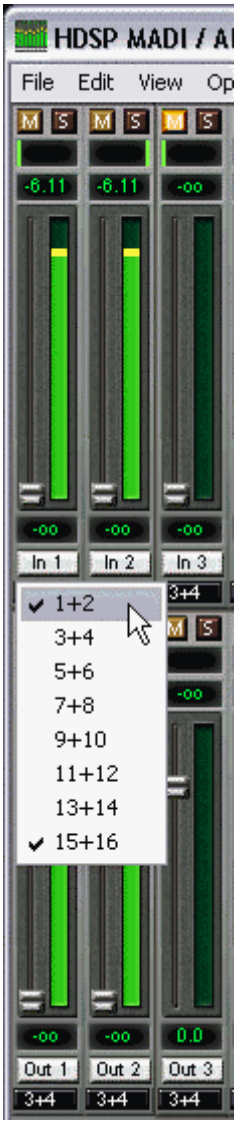
Hold down Ctrl and click on preset button 1 to make sure that factory preset 1 is loaded. The faders in the top row are set to maximum attenuation (called m.a. in the following), so there is no monitoring of the input channels. The **Submix View** is active, therefore for improved overview all outputs except 1/2 are greyed out. Additionally all faders are set to the current routing target 1/2.

We will now create a submix on outputs 3+4. Please start a multitrack playback. In the third row, click on the channels of hardware output 3 or 4. The Submix View changes to 3/4. Both the fader settings and the output levels of all other channels are still visible, but greyed out for improved orientation.

As soon as 3/4 became active, all faders of the second row jumped to their bottom position. This is correct, because as mentioned above the factory preset includes a 1:1 routing. Click on 1/2 and the faders above are the only active ones, same for 5/6 and so on.

Back to 3/4. Now you can change all the faders of all inputs and playback channels just as you like, thus making any input and playback signals audible via the outputs 3/4. The panorama can be changed too. Click into the area above the fader and drag the green bar in order to set the panorama between left and right. The level meters of the third row display the level changes in real-time.

As shown it is very easy to set up a specific submix for whatever output: select output channel, set up fader and pans of inputs and playbacks – ready!



For advanced users sometimes it makes sense to work without Submix View. Example: you want to see and set up some channels of different submixes simultaneously, without the need to change between them all the time. Switch off the Submix View by a click on the green button. Now the black routing fields below the faders no longer show the same entry (3+4), but completely different ones. The fader and pan position is the one of the individually shown routing destination.

In playback channel 1 (middle row), labeled Out 1, click onto the routing field below the label. A list pops up, showing a checkmark in front of '1+2'. So currently playback channel 1 is sent to this routing destination. Click onto '7+8'. The list disappears, the routing field no longer shows '1+2', but '7+8'. 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 channel 7, you can see the level of what you are hearing from output 7. 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 7 and 8. 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 output 8 as well.

In short: While editing the Submix 7+8 you have direct access to other submixes on other channels, because their routing fields are set to different destinations. And you get a direct view of how their faders and panoramas are set up.

This kind of visual presentation is very powerful, but for many users hard to understand, and requires a deep understanding of complex routing visualizations. Therefore we usually re-commend to work in ***Submix View****.*

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. Hold down the Ctrl-key and click into the routing field of Out 3. The routing list pops up with a checkmark at '3+4'. Select '7+8'. Now, Out 4 has already been set to '7+8' 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 field. You'll now see two checkmarks, at '3+4' and '7+8'. Click onto '9+10'. The window disappears, fader and pan jump to m.a., the signal can now be routed to the outputs 9 and 10. You can continue like this 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 signal at the outputs 7/8 did not change while you were routing channel 4 to other outputs and setting different gain values for those. 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 (stored) setting as soon as another routing is chosen.

Sometimes you will want the routings not to be independent. Let's say you have sent a signal to several submixes, and now want to change the signal's volume a bit on *all* these submixes. Dragging the faders by use of the right mouse button activates **Post Send** mode and causes all routings of the current input or playback channel to be changed in a relative way. Please note that the fader settings of all routings are memorized. So when pulling the fader to the bottom (maximum attenuation), the individual settings are back when you right click the mouse and pull the fader up. The individual settings get lost in m.a. position as soon as the fader is clicked with the left mouse button. As long as no single level is at m.a. position, the left mouse button can be used to change the current routing's gain.

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 '7+8' in the routing window, pull the fader down, open the routing window again - the checkmark is gone.

The number of channels is reduced automatically when entering Double Wire or Quad Wore mode. The display is adjusted accordingly, and all fader settings remain stored.

24.5 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 32 completely independent stereo submixes, 16 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 solved by **Submix View** mode. In this mode, all routing fileds 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 '1+2'). 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.

Changing to a different destination (output channel) is done in any routing field, or by a click on the desired output pair in the bottom row.

24.6 Mute und Solo

Mute operates pre-fader, thus mutes all currently active routings of the channel. As soon as any Mute button is pressed, the *Mute Master* button lights up in the Quick Access area. It allows to switch all selected mutes off and on again. You can comfortably make mute-groups or activate and deactivate several Mutes simultaneously.

The same holds true for the Solo and the *Solo Master* buttons. As with conventional mixing desks, Solo operates only for the output defined as **Monitor Main**, as a solo-in-place, post fader. As soon as one Solo button is pressed, the *Solo Master* button lights up in the Quick Access area. It allows to switch all selected Solos off and on again. You can comfortably make solo-groups or activate and deactivate several Solos simultaneously.

24.7 The Quick Access Panel

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

In the **View** section the single mixer 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 window can thus be reduced to the playback channels to save space. All combinations are possible and allowed.

As described earlier, **Submix** sets all routing windows to the same selection. Deactivating Submix automatically recalls the previous view. The mixer can be made smaller horizontally and vertically. This way TotalMix can be made substantially smaller and space-saving on the desktop/screen, if you have to monitor or set only a few channels or level meters.

The **Presets** are one of the most powerful and useful features of TotalMix. Behind the eight buttons, eight files are hidden (see next chapter). These contain the complete mixer state. 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 any 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 has been 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 have long file names.

Instead of single presets a complete bank of (8) presets can be loaded at once. Advantage: The names defined for the preset buttons will be stored and loaded automatically.

Up to three HDSP AES-32 and HDSP MADI can be used simultaneously. The **Unit** buttons switch between the cards. Holding down Ctrl while clicking on button Unit 2 or Unit 3 will open another TotalMix window.

24.8 Presets

TotalMix includes eight factory presets, stored within the program. The user presets can be changed at any time, because TotalMix stores and reads the changed presets from the files **preset11.mad** to **preset81.mad**, located in Windows' hidden directory *Documents and Settings, <Username>, Local Settings, Application Data, RME TotalMix*. On the Mac the location is in the folder *User, <Username>, Library / Preferences / Hammerfall DSP*. The first number indicates the current preset, the second number the current unit.

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 any time.

Mouse: The original factory presets 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.

Keyboard: Using Ctrl and any number between 1 and 8 (**not on the numeric keypad!**) will load the corresponding factory default preset. The key Alt will load the user presets instead.

When loading a preset file, for example 'Main Monitor AN 1_2 plus headphone mix 3_4.mad', the file name will be displayed in the title bar of the TotalMix window. Also when loading a preset by the preset buttons, the name of the preset is displayed in the title bar. This way it is always clear what the current TotalMix state is based on.

The eight factory presets offer a pretty good base to modify them to your personal needs. In all factory presets Submix View is active by default.

Preset 1
Description: All channels routed 1:1, monitoring of all playback channels.

Details: All inputs maximum attenuation. All playback channels 0 dB, routed to the same output. All outputs 0 dB. Level display set to RMS +3 dB. View Submix active.

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

Preset 2
Description: All channels routed 1:1, input and playback monitoring. As Preset 1, plus 1:1 pass through of all inputs.

Preset 3
Description: All channels routed 1:1, no input and no playback monitoring. All faders set to maximum attenuation.

Preset 4
Description: All channels routed 1:1, input and playback monitoring. As Preset 2, but all inputs muted.

Preset 5
Description: All channels routed 1:1, playback monitoring. Submix of all playback channels to channels 1/2.

Preset 6
Description: As preset 5, but submix of all input channels to channels 1/2.

Preset 7
Description: As preset 5, but submix of all input and playback channels to channels 1/2.

Preset 8
Description: Panic. As Preset 4, but playback channels muted too (no output signal).

Preset Banks
Instead of a single preset, all eight presets can be stored and loaded at once. This is done via Menu **File**, **Save All Presets as** and **Open All Presets** (file suffix .map). After the loading the presets can be activated by the preset buttons. In case the presets have been renamed (see chapter 24.11), these names will be stored and loaded too.

24.9 The Monitor Panel

The Monitor panel provides several options usually found on analog mixing desks. It offers quick access to monitoring functions which are needed all the time in typical studio work.

Monitor Main
Use the drop down menu to select the hardware outputs where your main monitors are connected to.

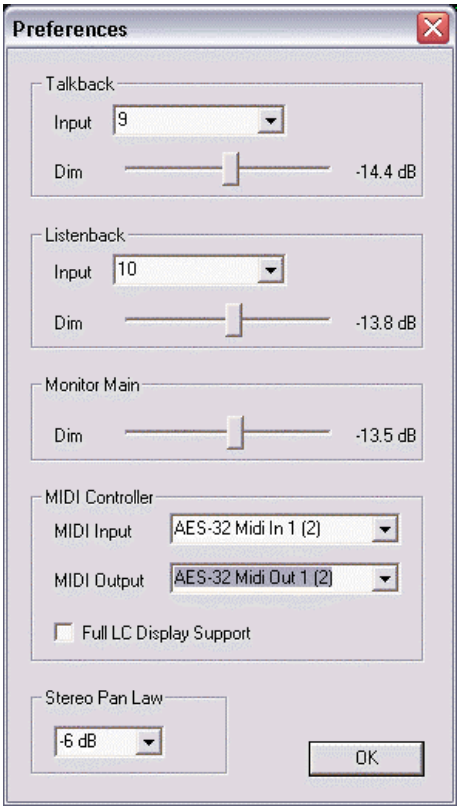
Dim
A click on this button will lower the volume of your main monitor output (see above) by an amount set up in the Preferences dialog (see below). This is the same as moving the third row faders down a bit, but much more convenient, as the old setting is back by a simple mouse click.

Mono
Sets the stereo output defined above to monaural playback. Useful to check for mono compatibility and phase problems.

Talkback
A click on this button will dim the Main Monitor output (see above) by an amount set up in the Preferences dialog. At the same time the control room's microphone signal (source defined in Preferences) is sent to the three destinations described below as *Monitor Phones*. The mic level is adjusted with the channel's input fader.

Monitor Phones 1/2/3
Use the drop down menu to select the hardware outputs where the submixes are sent to. These submixes are usually phones mixdowns for the musicians. A click on the button allows to hear the specific submix via the Main Monitor outputs. So when setting up or modifying the submix for the musician this process can be monitored easily and any time. Or in other words: you can easily check other hardware outputs/submixes by using the Monitor Phones function, without the need to copy/paste routings back and forth, or to reconfigure the cabling at the hardware.

24.10 Preferences



The dialog box Preferences is available via the menu *Options* or directly via F3.

Talkback
Input: Select the input channel of the Talkback signal (microphone in control room).
Dim: Amount of attenuation of the Main Monitor output in dB.

Listenback
Input: Select the input channel of the Listenback signal (microphone in recording room).
Dim: Amount of attenuation of the Monitor Phones outputs in dB.

Note: The Mute button of the Talkback and Listenback channel is still active. Therefore it is not necessary to select <NONE>, in case one of both shall be deactivated.

MIDI Controller, Full LC Display Support
See chapter 27.3 for details.

Main Monitor
Dim: Amount of attenuation of the Main Monitor output in dB. Activated by the *Dim* button in the Monitor panel.

Stereo Pan Law
The Pan Law can be set to -6 dB, -4.5 dB, -3 dB and 0 dB. The value chosen defines the level attenuation in pan center position. This setting is useful because the ASIO host often supports different pan laws too. Selecting the same value here and in the ASIO host, ASIO Direct Monitoring works perfectly, as both ASIO host and TotalMix use the same pan law. Of course, when not using ADM it can be changed to a setting different from the factory preset of –6 dB as well. You will most probably find that -3 dB gives a much more stable loudness when moving an object between left and right.

24.11 Editing the Names

The channel names shown in the white label area can be edited. A right mouse click on the white name field brings up the dialog box **Enter Name**. Any name can be entered in this dialog. Enter/Return closes the dialog box, the white label now shows the first letters of the new name. ESC cancels the process and closes the dialog box.

Moving the mouse above the label brings up a tool tip with the complete name.

The hardware outputs (third row) can be edited in the same way. In this case, the names in the routing drop down menus will change automatically. Additionally the names in the drop down menus of the Monitor section will change as well.

The preset buttons can get meaningful names in the same way. Move the mouse above a preset button, a right mouse click will bring up the dialog box. Note that the name shows up as tool tip only, as soon as the mouse stays above the preset button.

The preset button names are not stored in the preset files, but globally in the registry, so won't change when loading any file or saving any state as preset. But loading a preset bank (see chapter 24.8) the names will be updated.

24.12 Hotkeys

In many situations TotalMix can be controlled quickly and comfortably by the keyboard, making the mixer setup considerably easier and faster. The **Shift**-key for the fine mode for faders and panpots has already been mentioned. 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.
- 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. Alt plus number loads the user preset.
- Using multiple HDSP AES-32s, clicking the button *Unit 2* while holding down Ctrl opens a second TotalMix window for the second HDSP system, instead of replacing the window contents.

The faders can also be moved pairwise, corresponding to the stereo-routing settings. This is achieved by pressing the **Alt**-key and is especially comfortable when setting the Main Monitor and Phones submixes. Even the panoramas can be operated with Alt, from stereo through mono to inversed channels, and 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 current 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 for instance you want to set several faders to m.a., 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 of Input, Playback and Output channels each 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). The dialog box Preferences is opened via **F3**.

Hotkey **M** toggles Mute Master on/off (and with this performs a global mute on/off). Hotkey **X** toggles the Matrix view on/off (see chapter 25), hotkey **T** the mixer view. Hotkey **L** links all faders as stereo pairs.

Further hotkeys are available to control the configuration of the Level Meter (see chapter 24.14):

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

24.13 Menu Options

Always on Top: When active (checked) the TotalMix window 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 preset will not be used. The routing will be activated, but the window will not change.

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

Link Faders: Selecting this option all faders will be treated as stereo pairs and moved pairwise. Hotkey L.

MS Processing: Macro for a quick configuration of routing and phase for Mid/Side encoding and decoding. See chapter 26.7.

Level Meter Setup: Configuration of the Level Meters. Hotkey F2. See chapter 24.14.

Preferences: Opens a dialog box to configure several functions, like Pan Law, Dim, Talkback Dim, Listenback Dim. See chapter 24.10.

Enable MIDI Control: Turns MIDI control on.The channels which are currently under MIDI control are indicated by a colour change of the info field below the faders, black turns to yellow.

Deactivate MIDI in Background: Disables the MIDI control as soon as another application is in the focus, or in case TotalMix has been minimized.

24.14 Level Meter

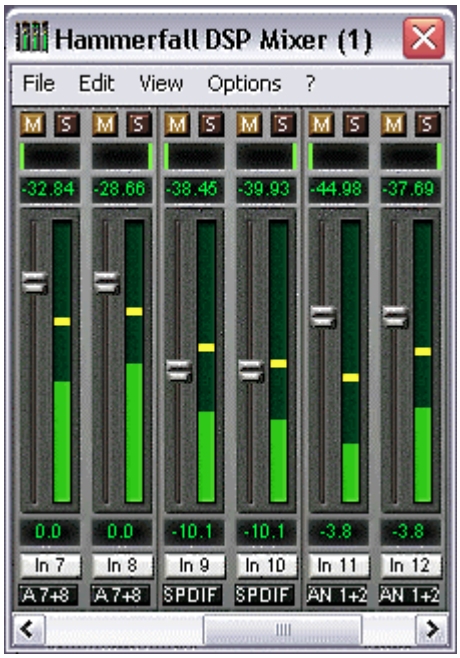
The HDSP AES-32 calculates all the display values Peak, Over and RMS in hardware, in order to be capable of using them independent of the software in use, and to significantly reduce the CPU load.

Tip: This feature, the **Hardware Level Meter**, is used by **DIGICheck** (Windows only, see chapter 16) to display Peak/RMS level meters of all channels, nearly without any CPU load.

The level meters integrated in TotalMix - considering their size - cannot be compared with DIGICheck. Nevertheless they already include many useful functions.

Peak and RMS is displayed for every channel. 'Level Meter Setup' (menu Options or F2) and direct keyboard entry (*hotkeys*) make 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. A RMS measurement shows 3 dB less for sine signals. While this is mathematically correct, it is not very reasonable for a level meter. Therefore the RMS readout is usually corrected by 3 dB, so that a full scale sine signal shows 0 dBFS on both Peak and RMS meters. This setting also yields directly readable signal-to-noise values. Otherwise the value shown with noise is 3 dB better than it actually is (because the reference is not 0 dB, but -3 dB). For example in WaveLab.

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 example: An *RME ADI-8 DS* connected to the HDSP AES-32 will show around -113 dBFS on all eight channel's input level meters.

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 the disappointing 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'.

25. TotalMix: The Matrix

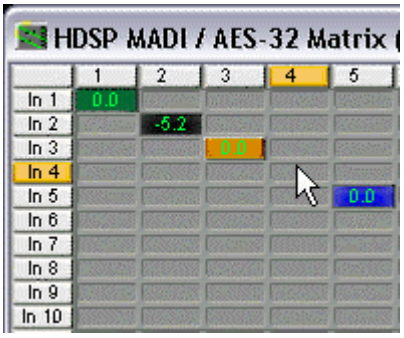
25.1 Overview

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 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.

25.2 Elements of the Matrix View

The visual 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 channels)
- 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.
- Blue field: Phase 180° - inverted.

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.

25.3 Operation

Using the Matrix is a breeze. It is very easy to indentify the current crosspoint, because the outer 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 **In 1** / **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 channels, and the upper side, representing the hardware outputs. Moving a fader in row 1 or 2 in TotalMix view, only the specific levels (max. 2) of this routing will change within 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.

A blue field indicates phase inversion. This state is displayed in the Matrix only, and can only be changed within the Matrix view. Hold down the Shift-key while clicking on an already activated field. Mute overwrites the phase display, blue becomes orange. If mute is deactivated the phase inversion is indicated again.

25.4 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 of 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 Phones output (equals factory preset 7). 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, not sending audio to a different output?

The most effective method to check a routing in mixer view is the Submix View, stepping through all existing software outputs, and having 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 TotalMix view how pan changes to achieve this special gain and routing when performing the second (fourth...) setting.

26. TotalMix Super-Features

26.1 ASIO Direct Monitoring (Windows only)

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 calculations, 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.

26.2 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: The relative positions are memorized until the faders are pulled down so that they reach upper or lower maximum position *and* the group is changed (select another channel or deselect one of the group).

Tip: Gang some submixes and watch all routing levels change in the Matrix view.

26.3 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 window open as second window 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 5/6, but now need the exact same signal also on the outputs 7/8. Click on Out 5, Ctrl-C, click on Out 7, Ctrl-V, same with 6/8 - 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.

26.4 Delete Routings

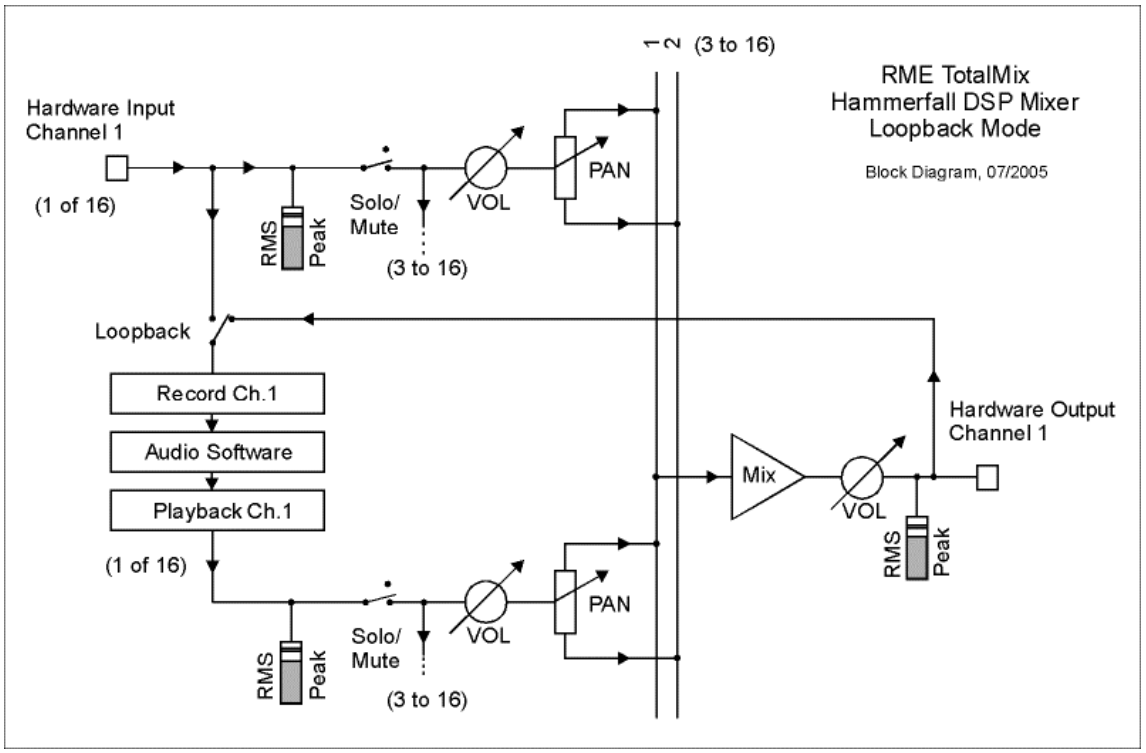
The fastest way to delete complex routings: select a channel in the mixer view, click on the menu entry *Edit* and select *Delete*. Or simply hit the Del-key. Attention: there is no undo in TotalMix, so be careful with this function!

26.5 Recording a Subgroup (Loopback)

TotalMix supports a routing of the subgroup outputs (=hardware outputs, bottom row) to the recording software. Instead of the signal at the hardware input, the signal at the hardware output is sent to the record software. This way, complete submixes can be recorded without an external loopback cable. Also the playback of a software can be recorded by another software.

To activate this function, click on the white label in the third row while holding down the Ctrl-key. The label's colour changes to red. In case the channel has already been part of a group, the colour will change from yellow to orange, signalling that the group functionality is still active for this channel.

In loopback mode, the signal at the hardware input of the corresponding channel is no longer sent to the recording software, but still passed through to TotalMix. Therefore TotalMix can be used to route this input signal to any hardware output. Using the subgroup recording, the input can still be recorded on a different channel.



As each of the (up to) 16 hardware outputs can be routed to the record software, and none of these hardware inputs gets lost, TotalMix offers an overall flexibility and performance not rivaled by any other solution.

Additionally the risk of feedbacks, a basic problem of loopback methods, is highly reduced, because the feedback can not happen within the mixer, but only when the audio software is switched into monitoring mode. The block diagram shows how the software's input signal is played back, and fed back from the hardware output to the software input. A software monitoring on the subgroup record channels is only allowed as long as the monitoring is routed in both software and TotalMix to a different channel than the active subgroup recording one.

Recording a Software's playback
In real world application, recording a software's output with another software will show the following problem: The record software tries to open the same playback channel as the playback software (already active), or the playback one has already opened the input channel which should be used by the record software.

This problem can easily be solved. First make sure that all rules for proper multi-client operation are met (not using the same record/playback channels in both programs). Then route the playback signal via TotalMix to a hardware output in the range of the record software, and activate this channel via Ctrl-mouse for recording.

Mixing several input signals into one record channel
In some cases it is useful to record several sources in only one track. For example when using two microphones when recording instruments and loudspeakers. TotalMix' Loopback mode saves an external mixing desk. Simply route/mix the input signals to the same output (third row), then re-define this output into a record channel via Ctrl-mouse – that's it. This way any number of input channels from different sources can be recorded into one single track.

26.6 Using external Effects Devices

With TotalMix a usage of external hardware - like effects devices - is easy and flexible.

Example 1: The singer (microphone input channel 1) shall have some reverb on his headphones (outputs 11/12). A direct routing In 1 to Out 11/12 for monitoring had been set up already. The external reverb is connected to a free output, for example channel 8. In active mode Submix View click on channel 8 in the bottom row. Drag the fader of input 1 to about 0 dB and the panorama fully to the right. Adjust the input level at the reverb unit to an optimal setting. Next the output of the reverb unit is connected to a free stereo input, for example 5/6. Use the TotalMix level meters to adjust a matching output level at the reverb unit. Now click on channels 11/12 in the bottom row, and move the fader of inputs 5/6 until the reverb effect gets a bit too loud in the headphones. Now click on channel 8 in the bottom row again and drag fader 1 down a bit until the mix of original signal and reverb is perfect for the singer.

The described procedure is completely identical to the one when using an analog mixing desk. There the signal of the singer is sent to an output (usually labeled Aux), from there to a reverb unit, sent back from the reverb unit as stereo wet signal (no original sound), back in through a stereo input (e.g. Effect return) and mixed to the monitoring signal. The only difference: The Aux sends on mixing desks are post-fader. Changing the level of the original signal causes a change of the effects level (here the reverb) too, so that both always have the same ratio.

Tip: Such a functionality is available in TotalMix via the right mouse button! Dragging the faders by use of the right mouse button causes all routings of the current input or playback channel to be changed in a relative way. This completely equals the function **Aux post fader**.

Example 2: Inserting an effects device can be done as above, even within the record path. Other than in the example above the reverb unit also sends the original signal, and there is no routing of input 1 directly to outputs 11/12. To insert an effects device like a Compressor/Limiter directly into the record path, the input signal of channel 1 is sent by TotalMix to any output, to the Compressor, back from the Compressor to any input. This input is now selected within the record software.

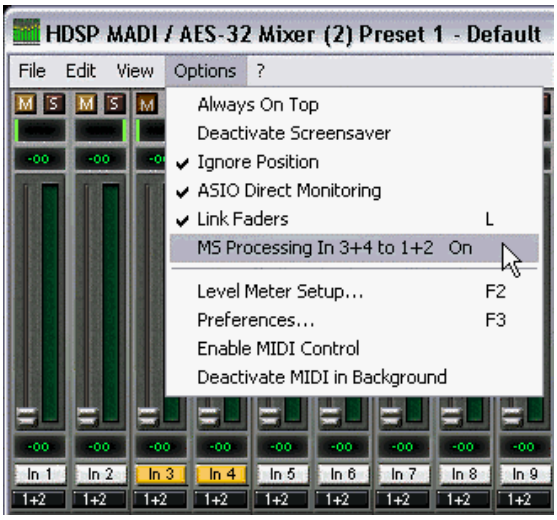
Unfortunately, very often it is not possible within the record software to assign a different input channel to an existing track 'on the fly'. The loopback mode solves this problem elegantly. The routing scheme stays the same, with the input channel 1 sent to any output via TotalMix, to the Compressor, from the Compressor back to any input. Now this input signal is routed directly to output 1, and output 1 is then switched into loopback mode via Ctrl-mouse.

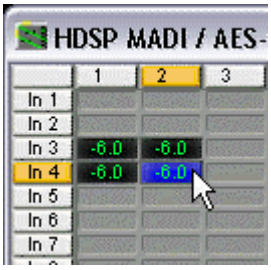
As explained in chapter 26.5, the hardware input of channel 1 now no longer feeds the record software, but is still connected to TotalMix (and thus to the Compressor). The record software receives the signal of submix channel 1 instead – the Compressor's return path.

26.7 MS Processing

The mid/side principle is a special positioning technique for microphones, which results in a mid signal on one channel and a side signal on the other channel. These information can be transformed back into a stereo signal quite easily. The process sends the monaural mid channel to left and right, the side channel too, but phase inverted (180°) to the right channel. For a better understanding: the mid channel represents the function L+R, while the side channel represents L-R.

During record the monitoring needs to be done in 'conventional' stereo. As TotalMix can invert the phase, it also offers the functionality of a M/S-decoder. The menu *Options* includes a macro to simplify the setup. First select the two input channels, in the picture to the right *Analog In 3* and *4*, having the current routing destination *Analog Out 1+2*. Now the string *MS Processing In 3+4 to AN 1+2 On* is shown in *Options*.





After a mouse click TotalMix sets gains and pans correctly. Of course these settings can also be performed manually. Repeat the last step to remove all routings (*menu Options ...Off*).

The M/S-Processing automatically operates as M/S encoder or decoder, depending on the source signal format. When processing a usual stereo signal, all monaural information will be shifted into the left channel, all stereo information into the right channel. Thus the stereo signal is M/S encoded. This yields some interesting insights into the mono/stereo contents of modern music productions. Additionally some very interesting methods of manipulating the stereo base and generating stereo effects come up, as it is then very easy to process the side channel with Low Cut, Expander, Compressor or Delay. The most basic application is already available directly in TotalMix: Changing the level of the side channel allows to manipulate the stereo width from mono to stereo up to extended, stepless and in real-time.

27. TotalMix MIDI Remote Control

27.1 Overview

TotalMix can be remote controlled via MIDI. It is compatible to the widely spread Mackie Control protocol, so TotalMix can be controlled with all hardware controllers supporting this standard. Examples are the Mackie Control, Tascam US-2400 or Behringer BCF 2000.

Additionally, the stereo output faders (lowest row) which are set up as *Monitor Main* outputs in the Monitor panel can also be controlled by the standard **Control Change Volume** via **MIDI channel 1**. With this, the main volume of the HDSP AES-32 is controlable from nearly any MIDI equipped hardware device.

27.2 Setup

- Open the Preferences dialog (menu Options or F3). Select the MIDI Input and MIDI Output port where your controller is connected to.

- When no feedback is needed (when using only standard MIDI commands instead of Mackie Control protocol) select NONE as MIDI Output.

- Check *Enable MIDI Control* in the Options menu.

27.3 Operation

The channels being under MIDI control are indicated by a colour change of the info field below the faders, black turns to yellow.

The 8-fader block can be moved horizontally and vertically, in steps of one or eight channels.

Faders can be selected to gang them.

In Submix View mode, the current routing destination (output bus) can be selected via REC Ch. 1 – 8. This equals the selection of a different output channel in the lowest row by a mouse click when in Submix View. In MIDI operation it is not necessary to jump to the lowest row to perform this selection. This way even the routing can be easily changed via MIDI.

Full LC Display Support: This option in Preferences (F3) activates complete Mackie Control LCD support with eight channel names and eight volume/pan values.

Attention: this feature causes heavy overload of the MIDI port when ganging more than 2 faders! In such a case, or when using the Behringer BCF2000, turn off this option.

When *Full LC Display Support* is turned off, only a brief information about the first fader of the block (channel and row) is sent. This brief information is also available on the LED display of the Behringer BCF2000.

Tip for Mac OS X users: LC Xview (*www.opuslocus.com*) provides an on-screen display emulating the hardware displays of a Logic/Mackie Control, for use with controllers that can emulate a Logic/Mackie Control but do not have a display. Examples include the Behringer BCF2000 and Edirol PCR series.

Deactivate MIDI in Background (menu Options) disables the MIDI control as soon as another application is in the focus, or in case TotalMix has been minimized. This way the hardware controller will control the main DAW application only, except when TotalMix is in the foreground. Often the DAW application can be set to become inactive in background too, so that MIDI control is switched between TotalMix and the application automatically when switching between both applications.

TotalMix also supports the 9th fader of the Mackie Control. This fader (labeled Master) will control the stereo output faders (lowest row) which are set up as *Main Monitor* outputs in the Monitor panel. Always and only.

27.4 Mapping

TotalMix supports the following Mackie Control surface elements*:

Element:	**Meaning in TotalMix:**
Channel faders 1 – 8	volume
Master fader	Main Monitor channel's faders
SEL(1-8) + DYNAMICS	reset fader to Unity Gain
V-Pots 1 – 8	pan
pressing V-Pot knobs	pan = center
CHANNEL LEFT or REWIND	move one channel left
CHANNEL RIGHT or FAST FORWARD	move one channel right
BANK LEFT or ARROW LEFT	move eight channels left
BANK RIGHT or ARROW RIGHT	move eight channels right
ARROW UP or Assignable1/PAGE+	move one row up
ARROW DOWN or Assignable2/PAGE-	move one row down
EQ	Master Mute
PLUGINS/INSERT	Master Solo
STOP	Dim Main Monitor
PLAY	Talkback
PAN	Mono Main Monitor
MUTE Ch. 1 – 8	Mute
SOLO Ch. 1 – 8	Solo
SELECT Ch. 1 – 8	Select
REC Ch. 1 – 8	in Submix mode only: select output bus
F1 - F8	load preset 1 - 8
F9	select Main Monitor
F10 - F12	Monitor Phones 1 - 3

*Tested with Behringer BCF2000 Firmware v1.07 in Mackie Control emulation for Steinberg mode and with Mackie Control under Mac OS X.

27.5 Simple MIDI Control

The stereo output faders (lowest row) which are set up as *Monitor Main* outputs in the Monitor panel can also be controlled by the standard **Control Change Volume** via **MIDI channel 1**. With this, the main volume of the HDSP AES-32 is controlable from nearly any MIDI equipped hardware device.

Even if you don't want to control all faders and pans, some buttons are highly desired to be available in 'hardware'. These are mainly the *Talkback* and the *Dim* button, and the new monitoring options (listen to Phones submixes). Fortunately a Mackie Control compatible controller is not required to control these buttons, as they are steered by simple Note On/Off commands on MIDI channel 1.

The notes are (hex / decimal / keys):

Monitor Main: 3E / 62 / **D 3**
Dim: 5D / 93 / **A 5**
Mono: 2A / 42 / **#F 1**
Talkback: 5E / 94 / **#A 5**

Monitor Phones 1: 3F / 63 / **#D 3**
Monitor Phones 2: 40 / 64 / **E 3**
Monitor Phones 3: 41 / 65 / **F 3**

Preset 1: 36 / 54 / **#F 2**
Preset 2: 37 / 55 / **G 2**
Preset 3: 38 / 56 / **#G 2**
Preset 4: 39 / 57 / **A 2**
Preset 5: 3A / 58 / **#A 2**
Preset 6: 3B / 59 / **B 2**
Preset 7: 3C / 60 / **C 3**
Preset 8: 3D / 61 / #**C 3**

An example of a small MIDI controller covering such MIDI functionality (and even some more) is the **Behringer BCN44**. This little box has 4 pots and 8 buttons for all the above functions – for less than 60 Euros.

27.6 Loopback Detection

The Mackie Control protocol requires feedback of the received commands, back to the hardware controller. So usually TotalMix will be set up with both a MIDI input and MIDI output. Unfortunately any small error in wiring and setup will cause a MIDI feedback loop here, which then completely blocks the computer (the CPU).

To prevent the computer from freezing, TotalMix sends a special MIDI note every 0.5 seconds to its MIDI output. As soon as it detects this special note at the input, the MIDI functionality is disabled. After fixing the loopback, check *Enable MIDI Control* under Options to reactivate the TotalMix MIDI.

User's Guide

HDSP AES-32

Technical Reference

28. 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 web-site, 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.

Driver updates Hammerfall DSP – Lists all changes of the driver updates.

DIGICheck: Analysis, tests and measurements with RME audio hardware
A description of DIGICheck, including technical background information.

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

Many background information on laptops and tests of notebooks:
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

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

29. Technical Specifications

29.1 Inputs

AES/EBU

- 8 x, transformer balanced, ground-free, according to AES3-1992
- High-sensitivity input stage (< 0.3 Vpp)
- SPDIF compatible (IEC 60958)
- Accepts Consumer and Professional format, copy protection will be ignored
- Single Wire: 8 x 2 channels 24 bit, up to 192 kHz
- Double Wire: 8 x 2 channels 24 bit 96 kHz, equalling 8 channels 192 kHz
- Quad Wire: 8 x 2 channels 24 bit 48 kHz, equalling 4 channels 192 kHz
- Lock range: 28 kHz - 204 kHz
- Jitter when synced to input signal: < 1 ns
- Jitter suppression: > 30 dB (2.4 kHz)

Word Clock

- BNC, not terminated (10 kOhm)
- Internal jumper for 75 Ohm termination
- Automatic Double/Quad Speed detection and conversion to Single Speed
- SteadyClock guarantees super low jitter synchronization even in varispeed operation
- Transformer coupled, galvanically isolated input
- Not effected by DC-offsets within the network
- Signal Adaptation Circuit: signal refresh through auto-center and hysteresis
- Overvoltage protection
- Level range: 1.0 Vss – 5.6 Vpp
- Lock range: 28 kHz – 204 kHz
- Jitter when synced to input signal: < 1 ns
- Jitter suppression: > 30 dB (2.4 kHz)

29.2 Outputs

AES/EBU

- 8 x, transformer balanced, ground-free, according to AES3-1992
- Output voltage Professional 4.5 Vpp, Consumer 2.1 Vpp
- Format Professional according to AES3-1992 Amendment 4
- Format Consumer (SPDIF) according to IEC 60958
- Single Wire: 8 x 2 channels 24 bit, up to 192 kHz
- Double Wire: 8 x 2 channels 24 bit 96 kHz, equalling 8 channels 192 kHz
- Quad Wire: 8 x 2 channels 24 bit 48 kHz, equalling 4 channels 192 kHz

Word Clock

- 1 x BNC
- Max. output voltage: 5 Vpp
- Output voltage @ 75 Ohm termination: 4.0 Vpp
- Output impedance: 10 Ohm
- Frequency range: 28 kHz – 204 kHz

29.3 Digital

- Clocks: Internal, AES/EBU In, Word Clock In, optional Video and LTC 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)
- Input PLL ensures zero dropout, even at more than 100 ns jitter
- Supported sample rates: 28 kHz up to 204 kHz

29.4 MIDI

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

29.5 Transfer Modes: Resolution / Bits per Sample

ASIO

- 24 / 32 bit 4 bytes (stereo 8 bytes)

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

MME

- 16 bit, 2 bytes (stereo 4 bytes)
- 20 bit, 3 bytes MSB (stereo 6 bytes)
- 20 bit, 4 bytes MSB (stereo 8 bytes)
- 24 bit, 3 bytes (stereo 6 bytes)
- 24 bit, 4 bytes MSB (stereo 8 bytes)
- 32 bit, 4 bytes (stereo 8 bytes)

8-channel interleaved:

- 16 bit, 2 bytes (stereo 4 bytes)
- 32 bit, 4 bytes (stereo 8 bytes)

30. Technical Background

30.1 AES/EBU - SPDIF

The most important electrical properties of 'AES' and 'SPDIF' can be seen in the table below. AES/EBU is the professional balanced connection using XLR plugs. The standard is being set by the *Audio Engineering Society* based on the AES3-1992. For the 'home user', SONY and Philips have omitted the balanced connection and use either Phono plugs or optical cables (TOSLINK). The format called S/P-DIF (SONY/Philips Digital Interface) is described by IEC 60958.

Type	**AES3-1992**	**IEC 60958**
Connection	XLR	RCA / Optical
Mode	Balanced	Un-balanced
Impedance	110 Ohm	75 Ohm
Level	0.2 V up to 5 Vss	0.2 V up to 0.5 Vss
Clock accuracy	not specified	I: ± 50ppm II: 0,1% III: Variable Pitch
Jitter	< 0.025 UI (4.4 ns @ 44.1 kHz)	not specified

Besides the electrical differences, both formats also have a slightly different setup. The two formats are compatible in principle, because the audio information is stored in the same place in the data stream. However, there are blocks of additional information, which are different for both standards. In the table, the meaning of the first byte (#0) is shown for both formats. The first bit already determines whether the following bits should be read as Professional or Consumer information.

Byte	**Mode**	**Bit 0**	**1**	**2**	**3**	**4**	**5**	**6**	**7**
0	Pro	P/C	Audio?	Emphasis			Locked	Sample Freq.	
0	Con	P/C	Audio?	Copy	Emphasis			Mode	

It becomes obvious that the meaning of the following bits differs quite substantially between the two formats. If a device like a common DAT recorder only has an SPDIF input, it usually understands only this format. In most cases, it will switch off when being fed Professional-coded data. The table shows that a Professional-coded signal would lead to malfunctions for copy prohibition and emphasis, if being read as Consumer-coded data.

Nowadays many devices with SPDIF input can handle Professional subcode. Devices with AES3 input almost always accept Consumer SPDIF (passive cable adapter necessary).

30.2 Lock and SyncCheck

Digital signals consist of a carrier and the data. If a digital signal is applied to an input, the receiver has to synchronize to the carrier clock in order to read the data correctly. To achieve this, the receiver uses a PLL (Phase Locked Loop). As soon as the receiver meets the exact frequency of the incoming signal, it is locked. This **Lock** state remains even with small changes of the frequency, because the PLL tracks the receiver's frequency.

If a AES/EBU signal is applied to the HDSP AES-32, the unit indicates LOCK, i. e. a valid input signal. This information is presented in the HDSP AES-32's Settings dialog. In the status display *SyncCheck,* the state of all clocks is decoded and shown as simple text (No Lock, Lock, Sync).

Unfortunately, LOCK does not necessarily mean that the received signal is correct with respect to the clock which processes the read out of the embedded data. Example: The HDSP AES-32 is set to 44.1 kHz internally (clock mode Master), and a mixing desk with AES output is connected to the card's AES1 input. The status display will show LOCK immediately, but usually the mixing desk's sample rate is generated internally (it is Master too), and thus slightly higher or lower than the HDSP AES-32's internal sample rate. Result: When reading out the data, there will frequently be read errors that cause clicks and drop outs.

Also when using multiple clock signals, a simple LOCK is not sufficient. The above described problem can be solved elegantly by setting the HDSP AES-32 from Master to AutoSync (its internal clock will then be the clock delivered by the mixing desk). But in case the card is clocked to word clock, this signal can also be un-synchronous, and there will again be a slight difference in the sample rate, and therefore clicks and drop outs.

In order to display those problems, the HDSP AES-32 includes **SyncCheck**®. It checks all clocks used for *synchronicity*. If they are not synchronous to each other, the status display will show LOCK. If they are synchronous to each other (i. e. absolutely identical), the status display will change to SYNC. In the example above it would have been obvious immediately that the entry LOCK is shown in *SyncCheck* instead of SYNC, right after connecting the mixing desk. With external synchronisation via word clock, both entries Word Clock and AESx must display SYNC.

In practice, SyncCheck allows for a quick overview of the correct configuration of all digital devices. So one of the most difficult and error-prone topics of the digital studio world finally becomes easy to handle.

A special problem occurs with devices offering several AES or SPDIF inputs. While with ADAT and TDIF all eight channels share the same clock base, with AES there are several completely independant receivers with their own PLLs and data buffers. Therefore there can be a random error of ± 1 sample difference between the stereo pairs. The HDSP AES-32's exclusive **SyncAlign**® technology avoids this effect and guarantees sample synchronicity among all four stereo channels.

30.3 Latency and Monitoring

The term **Zero Latency Monitoring** has been introduced by RME in 1998 for the DIGI96 series of audio cards. It stands for the ability to pass-through the computer's input signal at the interface directly to the output. Since then, the idea behind has become one of the most important features of modern hard disk recording. In the year 2000, RME published two ground-breaking Tech Infos on the topics *Low Latency Background*, which are still up-to-date: *Monitoring, ZLM and ASIO*, and *Buffer and Latency Jitter*, both found on the RME Driver CD and the RME website.

How much Zero is Zero?
From a technical view there is no zero. Even the analog pass-through is subject to phase errors, equalling a delay between input and output. However, delays below certain values can subjectively be claimed to be a zero-latency. This applies to analog routing and mixing, and in our opinion also to RME's Zero Latency Monitoring. The term describes the digital path of the audio data from the input of the interface to its output. The digital receiver of the HDSP AES-32 can't operate un-buffered, and together with TotalMix and the output via the transmitter, it causes a typical delay of 3 samples. At 44.1 kHz this equals about 68 µs (0.000068 s). In Double Speed mode, the delay doubles to 6 samples, for both ADAT and SPDIF.

Oversampling
While the delays of digital interfaces can be disregarded altogether, the analog inputs and outputs do cause a significant delay. Modern converter chips operate with 64 or 128 times oversampling plus digital filtering, in order to move the error-prone analog filters away from the audible frequency range as far as possible. This typically generates a delay of one millisecond. A playback and re-record of the same signal via DA and AD (loopback) then causes an offset of the newly recorded track of about 2 ms.

Buffer Size (Latency)
Windows: This option found in the Settings dialog defines the size of the buffers for the audio data used in ASIO and GSIF (see chapter 13 and 14).

Mac OS X: The buffer size is defined within the application. Only some do not offer any setting. For example iTunes is fixed to 512 samples.

General: A setting of 64 samples at 44.1 kHz causes a latency of 1.5 ms, for record and playback each. But when performing a digital loopback test no latency/offset can be detected. The reason is that the software naturally knows the size of the buffers, therefore is able to position the newly recorded data at a place equalling a latency-free system.

AD/DA Offset under ASIO and OS X: ASIO (Windows) and Core Audio (Mac OS X) allow for the signalling of an offset value to correct buffer independent delays, like AD- and DA-conversion or the Safety Buffer described below. An analog loopback test will then show no offset, because the application shifts the recorded data accordingly. Because in real world operation analog record and playback is unavoidable, the drivers include an offset value matching the HDSP AES-32's converter delays.

Because the HDSP AES-32 is a completely digital interface, and the delays introduced by external AD/DA-converters or other digital interfaces are unknown to unit and driver, the drivers include the digital offset values (3 / 6 samples). Therefore the delays caused by external converters have to be taken care off in the record software, which usually means that the user has to enter specific offset values manually.

Note: Cubase and Nuendo display the latency values signalled from the driver separately for record and playback. While with our digital cards these values equal exactly the buffer size (for example 3 ms at 128 samples), the HDSP AES-32 displays an additional millisecond – the time needed for the AD/DA-conversion.

Core Audios Safety Offset
Under OS X, every audio interface has to use a so called *satety offset*, otherwise Core Audio won't operate click-free. The HDSP AES-32 uses a safety offset of 32 samples. This offset is signalled to the system, and the software can calculate and display the total latency of buffer size plus AD/DA offset plus safety offset for the current sample rate.

30.4 DS - Double Speed

When activating the *Double Speed* mode the HDSP AES-32 operates at double sample rate. The internal clock 44.1 kHz turns to 88.2 kHz, 48 kHz to 96 kHz. The internal resolution is still 24 bit.

Sample rates above 48 kHz were not always taken for granted, and are still not widely used because of the CD format (44.1 kHz) dominating everything. Before 1998 there were no receiver/transmitter circuits available that could receive or transmit more than 48 kHz. Therefore a work-around was used: instead of two channels, one AES line only carries one channel, whose odd and even samples are being distributed to the former left and right channels. By this, you get the double amount of data, i. e. also double sample rate. Of course in order to transmit a stereo signal two AES/EBU ports are necessary then.

This transmission mode is called *Double Wire* in the professional studio world, and is also known as *S/MUX* (abbreviation for *Sample Multiplexing*) in connection with the multichannel ADAT format. The AES3 specification uses the uncommon term *Single channel double sampling frequency mode*.

Not before February 1998, Crystal shipped the first 'single wire' receiver/transmitters that could also work with double sample rate. It was then possible to transmit two channels of 96 kHz data via one AES/EBU port.

But *Double Wire* is still far from being dead. On one hand, there are still many devices which can't handle more than 48 kHz, e. g. digital tape recorders. But also other common interfaces like ADAT or TDIF are still using this technique. With MADI, sample multiplexing is often used as well to offer sample rates higher than 48 kHz.

The HDSP AES-32 supports all formats. 96 kHz can be received and transmitted both as 48K frame (using S/MUX) and as native 96K frame. In 48K frame Double Speed mode, the HDSP AES-32 distributes the data of one channel to two consecutive AES channels. This reduces the number of available channels from 16 to 8.

30.5 QS – Quad Speed

In earlier times the transmission of 192 kHz had not been possible via Single Wire, so once again sample multiplexing was used: instead of two channels, one AES line transmits only one half of a channel. A transmission of one channel requires two AES/EBU lines, stereo requires even four. This transmission mode is being called *Quad Wire* in the professional studio world. The AES3 specification does not mention Quad Wire.

The HDSP AES-32 supports all formats. 192 kHz can be received and transmitted as 48K frame (Quad Wire), 96K frame (Double Wire), and as native 192K frame. In 48K frame Quad Speed mode, the HDSP AES-32 distributes the data of one channel to four consecutive AES channels. This reduces the number of available channels from 16 to 4.

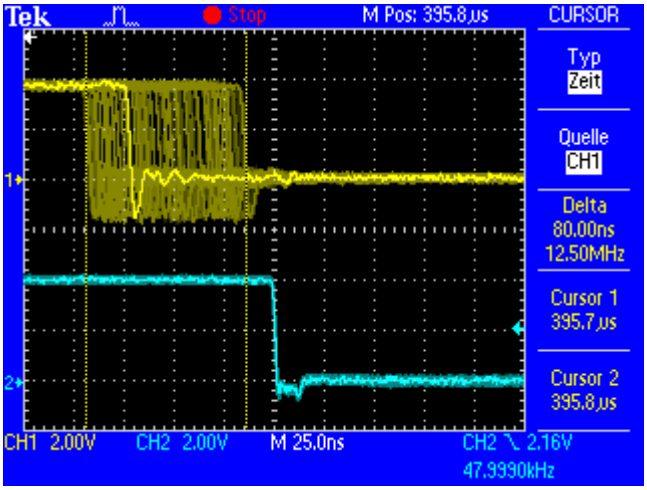
30.6 SteadyClock

The SteadyClock technology of the HDSP AES-32 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 the word clock output.

Usually a clock section consists of an analog PLL for external synchronization and several quartz oscillators for internal synchronisation. SteadyClock requires only one quartz, using a frequency not equalling digital audio. Latest circuit designs like hi-speed digital synthesizer, digital PLL, 100 MHz sample rate and analog filtering allow RME to realize a completely newly developed clock technology, right within the FPGA at lowest costs. The clock's performance exceeds even professional expectations. Despite its remarkable features, SteadyClock reacts quite fast compared to other techniques. It locks in fractions of a second to the input signal, follows even extreme varipitch changes with phase accuracy, and locks directly within a range of 28 kHz up to 200 kHz.

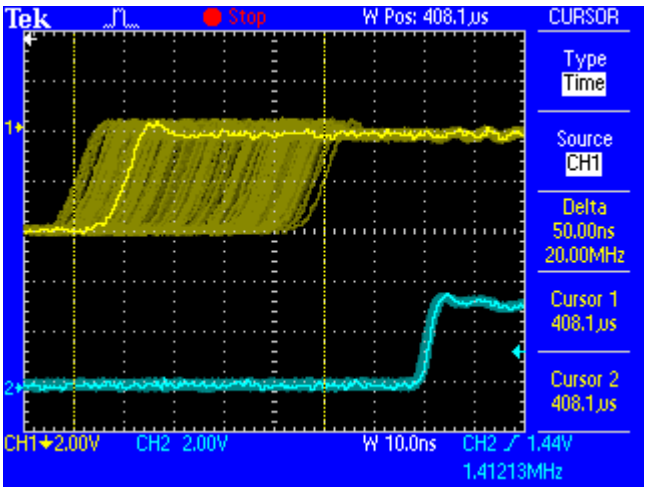
SteadyClock has originally been developed to gain a stable and clean clock from the heavily jittery MADI data signal. The embedded MADI clock suffers from about 80 ns jitter, caused by the time resolution of 125 MHz within the format. Common jitter values for other devices are 5 ns, while a very good clock will have less than 2 ns.

The picture to the right shows the MADI input signal with 80 ns of jitter (top graph, yellow). Thanks to SteadyClock this signal turns into a clock with less than 2 ns jitter (lower graph, blue).



The other input sources of the HDSP AES-32, AES/EBU, word clock, Video and LTC, gain a lot from SteadyClock as well. In fact, extracting a low jitter clock from LTC is not possible without a SteadyClock similar technique at all!

The scrnshot to the right shows an extremely jittery AES/EBU signal of about 50 ns jitter (top graph, yellow). Again SteadyClock provides an extreme clean-up. The filtered clock shows 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 HDSP AES-32 's word clock outputs. It is also used to clock the AES/EBU outputs.

30.7 PCI – Performance and Bus Load

The HDSP AES-32 card's sheer number of audio channels makes it more demanding for a computer's PCI bus performance than any other audio card. Furthermore, measurements of pure data throughput are not sufficient for measuring realtime audio performance or compatibility. Large amounts of data may be transferred in short and fast bursts with small interruptions, which will result in a relatively high data rate when measured averaged, but audio signals will suffer clicks and dropouts because of the interruptions.

Theoretically, PCI can transfer up to 133 MByte/s. A single HDSP AES-32 card will cause about 6 MByte/s of traffic. Therefore up to three cards can be used in any modern computer without problems. 48 channels I/O is no real challenge anymore, even notebooks can often handle this.

On a standard computer with modern single PCI bus, about 100 audio channels can be transmitted per direction (record/playback). Exceeding this limit, any system acitivity - even outside the PCI bus - causes drop outs. Transferring these experiences to the HDSP AES-32 means that besides the number of channels the *bus load* has to be taken into account too. One channel at 96 kHz causes the same load to the system as two channels at 48 kHz! So at 192 kHz, a HDSP AES-32 causes a bus load equalling a MADI card with 64 channels I/O at 48 kHz!

The usage of multiple AES-32 in DS and QS operation can therefore be problematic. Some examples:

- 2 HDSP AES-32 will most likely not run stable at 192 kHz at full track count. 2 x 16 channels 192 kHz equal 2 x 64 channels at 48 kHz = 128 channels per direction.

- 2 HDSP AES-32 at 96 kHz should operate reliable at full channel count. 2 x 16 equals 2 x 32 = 64 channels per direction.

- 3 HDSP AES-32 at 96 kHz can't operate at full channel count (3 x 16 equals 3 x 32 = 96 channels per direction). This limit may be different for different computers.

Using many channels at high sample rates, the follwing should be noted:

- The PCI-bus should be kept free of other devices. This also applies to on-board components such as modems, USB devices or network adapters.

- If disturbances occur during hard drive activity: Modern SATA hard drives often feature extremely high peak data rates, which are unnecessary for audio playback and recording, but tend to disturb the PCI bus. In this case, taking the step back to PATA controllers and drives can turn out to be a big step forward. On Macs, keeping audio files on an external FireWire drive may be worthwile. Even FireWire 800 efficiently limits peak performance, but the average data rate will be sufficient for many audio tracks.

Using two HDSP AES-32 cards at 192 kHz, or three at 96 kHz, requires boards with a very high transfer rate between north bridge and south bridge, in order to also allow the processing of other data (hard drive...) in real-time. The 266 MByte/s of an Intel 875 chipset are simply not sufficient. The solution is found in motherboards with server chipsets that usually incorporate several separated PCI busses. One good example is the Tyan 8KW (S2885), which not only operates flawlessly with thre HDSP AES-32 cards, but also provides ample CPU power, thanks to Dual Opteron CPUs.

30.8 Terminology

Single Speed
Sample rate range originally used in Digital Audio. Typical applications are 32 kHz (digital radio broadcast), 44.1 kHz (CD), and 48 kHz (DAT).

Double Speed
Doubles the original sample rate range, in order to achieve higher audio quality and improved audio processing. 64 kHz is practically never used, 88.2 kHz is quite rare in spite of certain advantages. 96 kHz is a common format. Sometimes called **Double Fast**.

Quad Speed
Controversially discussed way of ensuring hi-end audio quality and processing by quadrupling the sample frequency. 128 kHz is non-existant, 176.4 kHz is rare, if at all then 192 kHz is used, e.g. for DVD Audio.

Single Wire
Standard audio data transfer, where the audio signal's sample rate is equal to the rate of the digital signal. Used from 32 to 192 kHz. Sometimes called **Single Wide**.

Double Wire
Before 1998 there were no receiver/transmitter circuits available that could receive or transmit more than 48 kHz. Higher sample rates were transferred by splitting odd and even bits across the L/R channels of a single AES connection. This provides for twice the data rate, and hence twice the sample rate. A stereo signal subsequently requires two AES/EBU ports.

The Double Wire method is an industry standard today, however it has a number of different names, like **Dual AES**, **Double Wide**, **Dual Line** and **Wide Wire**. The AES3 specification uses the uncommon term *Single channel double sampling frequency mode*. When used with the ADAT format, the term S/MUX is commonly used.

Double Wire not only works with Single Speed signals, but also with Double Speed. As an example, Pro Tools HD, whose AES receiver/transmitter only work up to 96 kHz, uses Double Wire to transmit 192 kHz. Four channels of 96 kHz turn into two channels of 192 kHz.

Quad Wire
Similar to Double Wire, with samples of one channel spread across four channels. This way single speed devices can transmit up to 192 kHz, but need two AES/EBU ports to transmit one channel. Also called **Quad AES**.

S/MUX
Since the ADAT hardware interface is limited to Single Speed, the Double Wire method is used for sample rates up to 96 kHz, but usually referred to as S/MUX (Sample Multiplexing). An ADAT port supports four channels this way. With MADI S/MUX is used as well, to transmit up to 96kHz although the 48K Frame format is used.

S/MUX4
The Quad Wire method allows to transmit two channels at up to 192 kHz via ADAT. The method is referred to as S/MUX4. With MADI S/MUX4 is used as well, to transmit up to 192 kHz although the 48K Frame format is used.

Note: All conversions of the described methods are lossless. The existing samples are just spread or re-united between the channels.

48K Frame
Describes the digital format used with sample rates below 50 kHz. Also used to describe the standard MADI format (which supports up to 64 channels at up to 48 kHz).

96K Frame
Describes the digital format used with sample rates between 64 kHz and 100 kHz. Also used to describe the special MADI format (which supports up to 32 channels at up to 96 kHz). The advantage of this format against 48K frame using S/MUX: the receiver can detect the real (double) sample rate on its own and immediately. With 48K Frame and S/MUX, the user has to set up the correct sample rate in all involved devices manually.

192K Frame
Describes the digital format used with sample rates between 128 kHz and 200 kHz. The advantage of this format against 48K frame using S/MUX: the receiver can detect the real (double) sample rate on its own and immediately. With 48K Frame and S/MUX, the user has to set up the correct sample rate in all involved devices manually. And wastes a lot of channels.

30.9 Connector Pinouts

The 25 pin D-sub connector provides all four AES inputs and outputs. The pinout uses the widely spread Tascam scheme, which is also used by Digidesign.

Tascam / Digidesign:

Signal	In 1/2+	In 1/2-	In 3/4+	In 3/4-	In 5/6+	In 5/6-	In 7/8+	In 7/8-
D-Sub	24	12	10	23	21	9	7	20

Signal	Out 1/2+	Out 1/2-	Out 3/4+	Out 3/4-	Out 5/6+	Out 5/6-	Out 7/8+	Out 7/8-
D-Sub	18	6	4	17	15	3	1	14

GND is connected to pins 2, 5, 8, 11, 16, 19, 22, 25. Pin 13 is not connected.

The Yamaha pinout is quite popular as well. When building a D-sub to D-sub adapter or connection cable, please make sure that the connectors are clearly labeled with *Tascam* and *Yamaha*. The cable can only be used when the Tascam side is connected to a Tascam connector, and the Yamaha side is connected to a Yamaha connector.

Yamaha:

Signal	In 1/2+	In 1/2-	In 3/4+	In 3/4-	In 5/6+	In 5/6-	In 7/8+	In 7/8-
D-Sub	1	14	2	15	3	16	4	17

Signal	Out 1/2+	Out 1/2-	Out 3/4+	Out 3/4-	Out 5/6+	Out 5/6-	Out 7/8+	Out 7/8-
D-Sub	5	18	6	19	7	20	8	21

GND is connected to pins 9, 10, 11, 12, 13, 22, 23, 24, 25.

The same is true for a direct adapter cable Tascam D-sub to Euphonix D-sub.

Euphonix:

Signal	In 1/2+	In 1/2-	In 3/4+	In 3/4-	In 5/6+	In 5/6-	In 7/8+	In 7/8-
D-Sub	15	2	4	16	18	5	7	19

Signal	Out 1/2+	Out 1/2-	Out 3/4+	Out 3/4-	Out 5/6+	Out 5/6-	Out 7/8+	Out 7/8-
D-Sub	21	8	10	22	24	11	13	25

GND is connected to pins 3, 6, 9, 12, 14, 17, 20, 23. Pin 1 is not connected.

AES/EBU

The XLR connectors are wired according to AES3-1992:

1 = GND (shield)
2 = Signal
3 = Signal

AES/EBU and SPDIF are biphase modulated signals, therefore polarity doesn't matter. Pins 2 and 3 are neither hot nor cold, they carry the same signal. But as AES3 uses a balanced transmission they are inverted in polarity.