

User's Guide



Fireface 800

The most powerful FireWire audio interface ever!

DVD
24 Bit / 192 kHz ✓ *ready*

TotalMix™



SyncAlign®

ZLM®

SyncCheck®



SteadyClock™



FireWire 800 / 400 Digital I/O System
10 + 16 + 2 Channels Analog / ADAT / SPDIF Interface
24 Bit / 192 kHz Digital Audio
56 x 28 Matrix Router
MIDI I/O
Stand-Alone Operation

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1. Introduction

Thank you for choosing the Fireface 800. This unique audio system is capable of transferring analog and digital audio data directly to a computer from practically any device. The latest Plug and Play technology guarantees a simple installation, even for the inexperienced user. The numerous unique features and well thought-out configuration dialog puts the Fireface 800 at the very top of the range of computer-based audio interfaces.

The package contains drivers for Windows 2000/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 that your Fireface 800 package contains each of the following:

- RME Driver CD
- Cable IEEE1394a (FW400), 4.5 m (15 ft)
- Power cord
- Manual
- 1 optical cable (TOSLINK), 2 m (6.6 ft)

3. System Requirements

- Windows 2000 SP4, Windows XP, Mac OS X (10.2.8 or higher)
- FireWire Interface: a free FireWire Port 400 (1394a) or 800 (1394b)

Note: Examples and detailed descriptions of suitable audio desktop systems can be found in the Tech Info *RME Reference PCs: Hardware recommendations*. Information on compatibility and performance of notebooks/laptops is included in RMEs Tech Infos about notebooks, *HDSP System – Notebook Basics and Tests*. A report about incompatible FireWire 800 controllers is found in the Tech Info *FireWire 800 Hardware – Compatibility Problems*.

4. Brief Description and Characteristics

- All settings can be changed in real-time
- Enhanced Mixed Mode: Analog, ADAT and SPDIF I/O simultaneously usable
- 8 buffer sizes/latencies available: 1,0 / 1,5 / 2,1 / 3 / 6 / 12 / 17 / 23 ms
- 8 channels 96 kHz/24 Bit Record/Playback via ADAT optical (S/MUX)
- Clock modes slave and master
- Automatic and intelligent master/slave clock control
- Unsurpassed Bitclock PLL (audio synchronization) in ADAT mode
- Word clock input and output
- Enhanced ZLM for latency-free submixes and perfect ASIO Direct Monitoring
- SyncAlign guarantees sample aligned and never swapping channels
- SyncCheck tests and reports the synchronization status of input signals
- 1 x MIDI I/O, 16 channels high-speed MIDI
- 1 x Hi-power headphone output
- DIGICheck DSP: Level meter in hardware, peak- and RMS calculation
- TotalMix: 1568 channel mixer with 42 Bit internal resolution

5. Hardware Installation

Desktop Computer

- Use the power cord to connect the Fireface with any suitable power outlet.
- Connect computer and Fireface using the supplied 6-pin FireWire cable (IEEE1394a). In case your computer does not have any FireWire port, PCI-cards providing multiple FireWire ports can be added easily. FireWire cards are available in your local computer shop.
- Power on the computer. Switch on the Fireface when Windows shows the desktop.

Notebook

- Use the power cord to connect the Fireface with any suitable power outlet.
- Connect computer and Fireface using the supplied 6-pin FireWire cable (IEEE1394a). In case your notebook does not have any FireWire port, CardBus-cards can be used to add them. In case your notebook does not have a 6-pin jack, or any supplied 6-pin to 4-pin adapter: such adapters are available in your local computer store, also as complete cables 4-pin to 6-pin.
- Power on the notebook. Switch on the Fireface when Windows shows the desktop.

6. Driver Installation

6.1 Windows 2000/XP

After the Fireface has been switched on, (see 5. Hardware Installation) the green arrow symbol appears in the task bar (dialog *Remove hardware*) because Windows has detected an external hardware.



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 **Fireface W2k** on the RME Driver CD.

Windows now installs the driver of the Fireface 800 and registers it as a new audio device in the system. After a reboot, the symbols of mixer and Settings dialog will appear in the task bar (see above). The red Host error LED extinguishes.

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!*

Possible reasons why a Fireface is not found automatically:

- The FireWire port is not active in the system (drivers of the FireWire PCI or CardBus card have not been installed)
- The FireWire cable is not, or not correctly inserted into the socket
- No power. After switching the Fireface on, at least the red Host error LED has to be lit.

6.2 Driver Update

RME's driver updates often include a new fireface.inf file. To prevent Windows 2000/XP from using an old fireface.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 Fireface 800/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.

6.3 Flash Update

The Flash Update Tool updates the firmware of the Fireface 800 to the latest version. It requires an already installed driver.

Start the program fireface_fut.exe. The Flash Update Tool displays the current revision of the Fireface's firmware, 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 (Verify Ok).

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

After the update the unit needs to be reset. This is done by powering down the Fireface for a few seconds. A reboot of the computer is not necessary.

When the update fails (status: failure), the unit'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.

6.4 Deinstalling the Drivers

A deinstallation of the Fireface'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. These entries can be removed from the registry by a software deinstallation request. This request can be found (like all deinstallation entries) in *Control Panel, Software*. Click on the entry 'RME Fireface'.

7. First Usage - Quick Start

Connect the TRS-jacks or the XLR connectors with the analog signal source. The input sensitivity of the rear inputs can be changed in the Settings dialog (Input Level), assuring the highest signal to noise ratio will be achieved. Try to achieve an optimum input level by adjusting the source itself. Raise the source's output level until the level meters in TotalMix reach about -3 dB.

The analog line inputs of the Fireface 800 can be used with +4 dBu and -10 dBV signals. The electronic input stage can handle balanced (XLR, TRS jacks) and unbalanced (TS jacks) input signals correctly.

The front inputs signal level can be optimized using the Fireface's Gain pots. A Signal LED and a Clip LED help to find the correct level adjustment.

The Fireface's digital outputs provide SPDIF (AES/EBU compatible) and ADAT optical signals at the corresponding ports.

On the analog playback side (the DA side), a coarse adjustment of the analog output level at the rear jacks is available in the Settings dialog (Output Level).

The output signals of channels 9/10 is available on the front. Their output level can be set using the VOL pot. This output is a very low impedance type, which can also be used to connect headphones.

The function *Store in Flash Memory* (Settings dialog) and *Flash current mixer state* (TotalMix) allow to store the current settings into the Fireface 800. The unit then remembers all settings, and loads these automatically when switched on. With this, the Fireface 800 can be used stand-alone after setting it up accordingly, replacing lots of dedicated devices (see chapter 13).

8. Configuring the Fireface

8.1 Settings dialog - General

Configuration of the Fireface 800 is done via its own settings dialog. The panel 'Settings' can be opened:

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



The mixer of the Fireface 800 (TotalMix) can be opened:

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

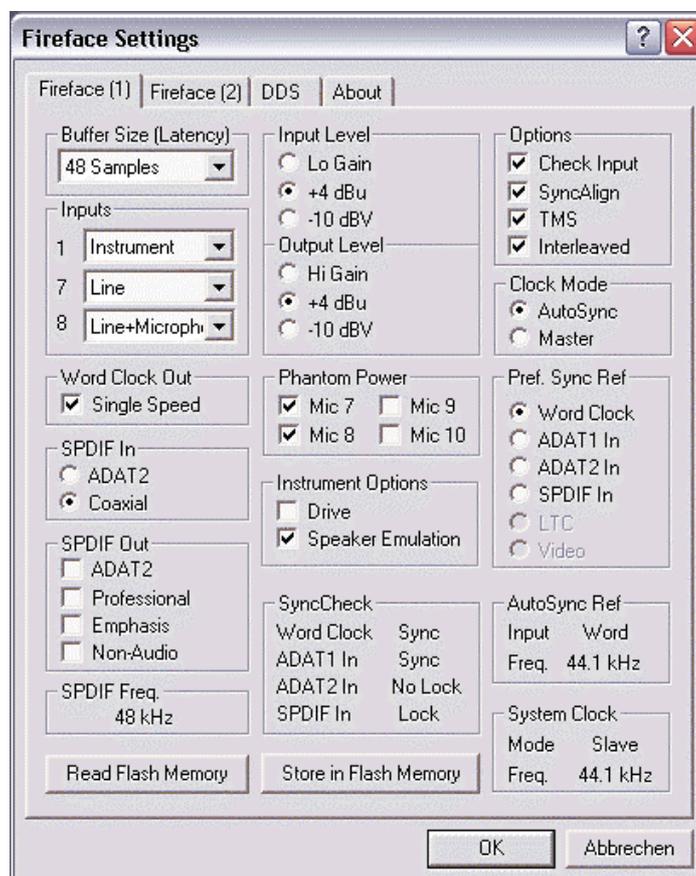


The hardware of the Fireface 800 offers a number of helpful, well thought-of practical functions and options which affect how the card operates - it can be configured to suit many different requirements. The following is available in the 'Settings' dialog:

- Input selection
- Level of analog I/Os
- 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. *Sync-Check* indicates whether there is a valid signal (Lock, No Lock) for each input (Word clock, ADAT1, ADAT2, SPDIF), or if there is a valid *and* synchronous signal (Sync). The *AutoSync Ref* display shows the input and frequency of the current sync source.



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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 11). Under Windows MME this setting determines the DMA buffer size (see chapter 9.4).

Inputs

Input selection for the channels 1, 7 and 8. Channel 1 can be the front Instrument input, or the rear TRS jack, or both simultaneously. Channel 7/8 can be the front microphone input, or the rear TRS jack, or both simultaneously.

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.

SPDIF In

Defines the input for the SPDIF signal. 'Coaxial' relates to the RCA socket, 'ADAT2' to the second optical TOSLINK input.

SPDIF Out

The SPDIF output signal is constantly available at the phono plug. After selecting 'ADAT2' it is also routed to the second optical TOSLINK output. For further details about the settings 'Professional', 'Emphasis' and 'Non-Audio', please refer to chapter 16.2.

SPDIF Freq.

Displays the sample rate of the SPDIF (or AES) input signal.

Input Level

Defines the reference level for the rear analog inputs 1-8.

Output Level

Defines the reference level for the rear analog outputs 1-8.

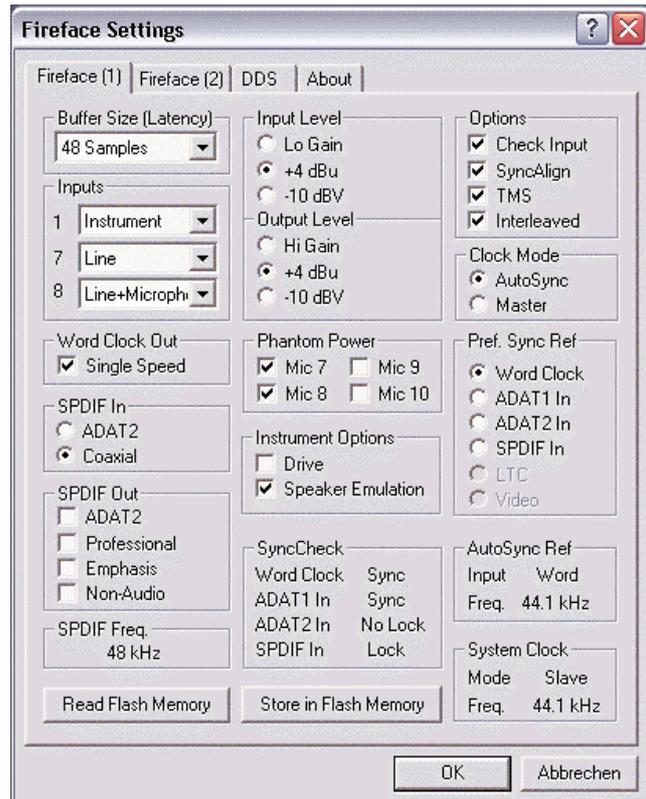
Phantom Power

Phantom power (48V) can be selected for each microphone input separately.

Instrument Options

Drive activates 25 dB additional gain for maximum sustain and brute distortion.

Speaker Emulation removes low frequency noise and cuts off higher frequencies.



SyncCheck

SyncCheck indicates whether there is a valid signal (Lock, No Lock) for each input (Word clock, ADAT1, ADAT2, SPDIF), or if there is a valid *and* synchronous signal (Sync). The *AutoSync Ref* display shows the input and frequency of the current sync source.

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. *Check Input* 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 of the SPDIF input.

With *Interleaved* activated, the WDM devices can also be used as 8-channel devices (see chapter 9.6).

Clock Mode

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

Pref. Sync Ref.

Used to pre-select the desired clock source. If the selected source isn't available, the system will change to the next available one. The current clock source and sample rate is displayed in the *AutoSyncRef* display.

The automatic clock selection checks and changes between the clock sources Word Clock, ADAT1, ADAT2, SPDIF, LTC and Video (LTC/Video when using the optional TCO module).

System Clock

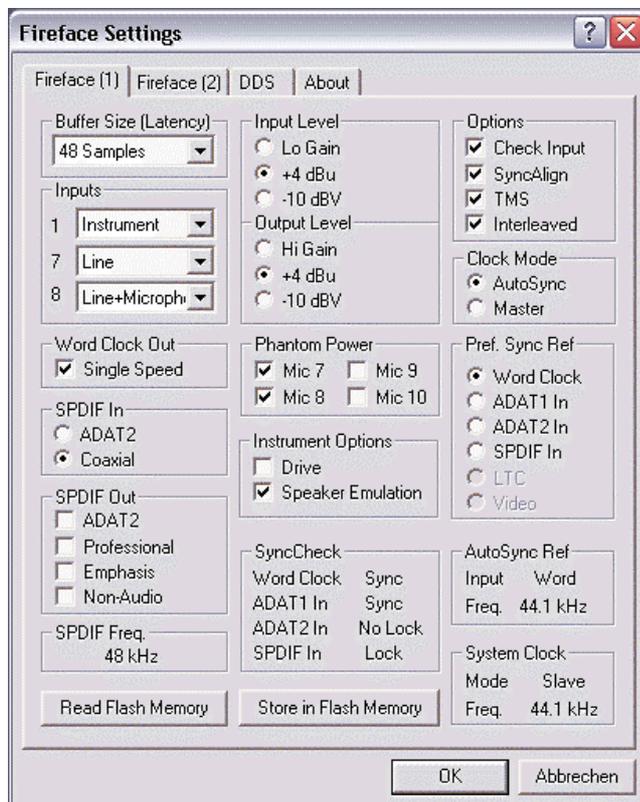
Shows the current clock state of the Fireface 800. The system is either Master (using its own clock) or Slave (AutoSync Ref).

Read Flash Memory

A click on this button causes all settings to change to the ones stored in the flash memory of the Fireface.

Store in Flash Memory

A click on this button transmits all current settings into the flash memory of the Fireface.



8.2 Settings dialog - DDS

Usually soundcards and audio interfaces get their internal clock (master mode) from 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 Fireface 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 Fireface to be in clock mode Master! The frequency setting will only be applied to the Fireface in this mode!*

 *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

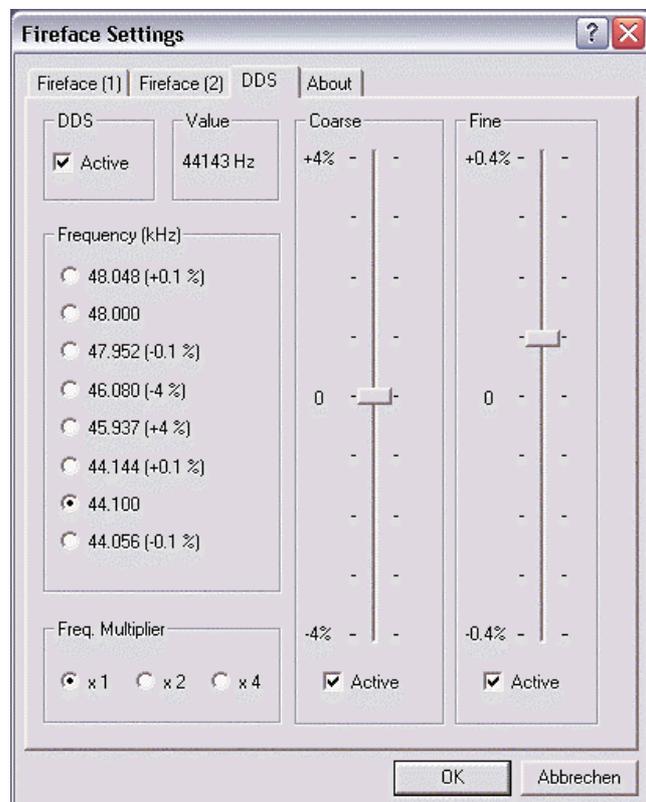
Changes the basic sample rate into Single, Double oder Quad Speed mode.

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 knob, 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 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 ADAT channels.

8.3 Clock Modes - Synchronization

In the digital world, all devices must be either 'Master' (clock source) or 'Slave' synchronized to a master. Whenever several devices are linked within a system, there must always be a single master clock. The Fireface's intelligent clock control is very user-friendly, being able to switch between clock modes automatically. Selecting **AutoSync** will activate this mode.

In AutoSync mode, the system constantly scans all digital inputs for a valid signal. If this signal corresponds with the current playback sample rate, the card switches from the internal quartz (System Clock Mode displays Master) to a clock generated from the input signal (System Clock Mode displays Slave). This allows on-the-fly recording, even during playback, without having to synchronize the card to the input signal first. It also allows immediate playback at any sample rate without having to reconfigure the unit.

AutoSync guarantees that normal record and record-while-play will always work correctly. In certain cases however, e.g. when the inputs and outputs of a DAT machine are connected directly to the Fireface 800, AutoSync may cause feedback in the digital carrier, so synchronization breaks down. To remedy this, switch the Fireface's clock mode over to 'Master'.



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

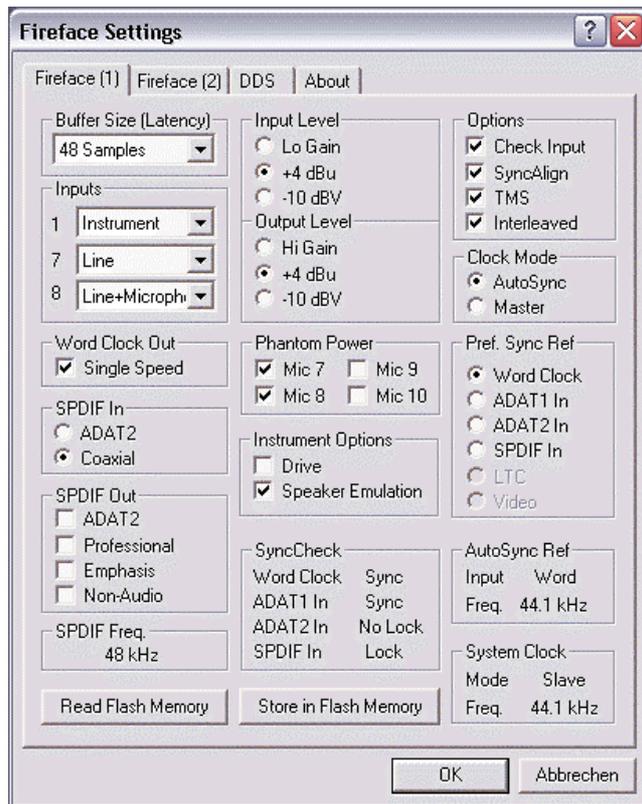
The Fireface's ADAT optical and SPDIF inputs operate simultaneously. Because there is no input selector however, the HDSP has to be told which of the signals is the sync reference (a digital device can only be clocked from a *single* source). The Fireface has been equipped with *AutoSync*, an automatic 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 sample frequency.

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

To cope with some situations which may arise in studio practice, setting 'Pref Sync Ref' is essential. One example: An ADAT recorder is connected to the ADAT input (ADAT immediately becomes the AutoSync source) and a CD player is connected to the SPDIF input. Try recording a few samples from the CD and you will be disappointed. Few CD players can be synchronized. The samples will inevitably be corrupted, because the signal from the CD player is read with the (wrong) clock from the ADAT i.e. out of sync. In this case, 'Pref Sync Ref' should be temporarily set to SPDIF.

If several digital devices are to be used simultaneously in a system, they not only have to operate with the same sample frequency but also be synchronous with each other. This is why digital systems always need a single device defined as 'master', which sends the same clock signal to all the other ('slave') devices. RME's exclusive **SyncCheck** technology (first implemented in the Hammerfall) enables an easy to use check and display of the current clock status. *SyncCheck* indicates whether there is a valid signal (Lock, No Lock) for each input (Word clock, ADAT1, ADAT2, SPDIF), or if there is a valid *and* synchronous signal (Sync). The *AutoSync Ref* display shows the input and frequency of the current sync source (see chapter 30.1).

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.



Thanks to its AutoSync technique and lightning fast PLLs, the Fireface 800 is not only capable of handling standard frequencies, but also any sample rate between 28 and 200 kHz. Even the word clock input, most often used in varispeed operation, allows any frequency between 28 kHz and 200 kHz.

At 88.2 or 96 kHz: If the ADAT input has been selected in *Pref Sync Ref*, the sample frequency shown in the *SPDIF In* field differs from the one shown in *AutoSync Ref*. The card automatically switches to its Sample Split mode here, because ADAT optical inputs and outputs are only specified up to 48 kHz. Data from/to a single input/output is spread over two channels, the internal frequency stays at 44.1 or 48 kHz. In such cases, the ADAT sample frequency is only half the SPDIF frequency.

9. Operation and Usage

9.1 Connectors

The front of the Fireface 800 features an Instrument input, microphone inputs and Line inputs with Gain pots, a stereo headphone output with volume pot, and several status LEDs.

MIDI/I indicates MIDI data received by the MIDI input.

MIDI/O indicates MIDI data sent to the MIDI output.

The **Clock Input State LEDs** (WC, SPDIF, ADAT, TCO) indicate separately for each digital input a valid input signal. Additionally, RME's exclusive *SyncCheck* indicates if one of these inputs is locked, but not synchronous to the others, in which case the LED will flash. See also chapter 8.3, Clock Modes - Synchronization.

The red **HOST** LED lights up when the Fireface 800 has been switched on, thus signalling the presence of operating voltage. At the same time it operates as Error LED, in case the FireWire connection hasn't been initialised yet, or has been interrupted (Error, cable not connected etc.).

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

The rear panel of the Fireface 800 features 8 analog inputs and outputs, the power socket, and all digital inputs and outputs:

ADAT1 I/O (TOSLINK)

ADAT2 I/O (TOSLINK): can also be used as optical SPDIF input and output, if set up accordingly in the Settings dialog.

SPDIF I/O coaxial (RCA): Fully AES/EBU compatible by transformer-coupling and level adjustment.

The SPDIF input is selected via the Settings dialog, started by clicking on the fire symbol in the Task Bar's system tray. The Fireface 800 accepts the commonly used digital audio formats, SPDIF as well as AES/EBU.

Wordclock I/O (BNC): A push switch allows to activate internal termination (75 Ohms).

IEC receptable for power connection. The specially developed, internal hi-performance switch mode power supply makes the Fireface operate in the range of 100V to 240V AC. It is absolutely short-circuit-proof, has an integrated line-filter, is fully regulated against voltage fluctuations, and suppresses mains interference.

9.2 Playback

The Fireface 800 can play back audio data in supported formats only (sample rate, bit resolution). Otherwise an error message appears (for example at 22 kHz and 8 bit).

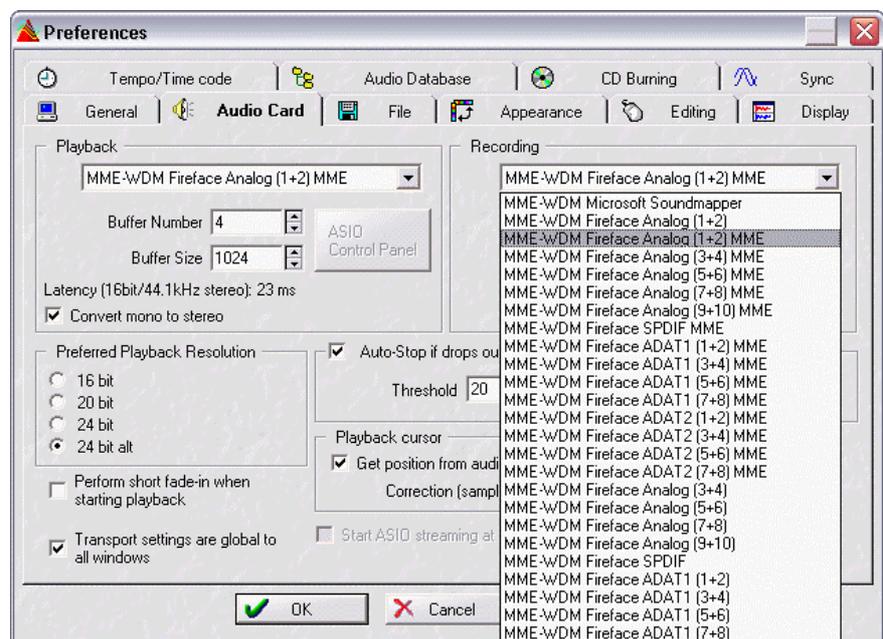
In the audio application being used, Fireface must be selected as output device. This can often be found in the *Options*, *Preferences* or *Settings* menus under *Playback Device*, *Audio Devices*, *Audio* etc.

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

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

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

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



The Fireface's ADAT optical interfaces support sample rates of up to 96 kHz using a standard ADAT recorder. Single-channel data at this frequency requires two ADAT channels, achieved using the Sample Split technique. This reduces the number of available ADAT channels from 8 to 4. Under Windows MME, channels are routed to ADAT devices in double-speed mode as follows:

- Only the 4 stereo pairs of the ADAT1 port are available
- The ADAT2 port is no longer available

This kind of implementation allows a problem-free usage of both ADAT ports in Single and Double Speed, as the routing for ADAT1 doesn't change. However, the hardware spreads the data differently:

- Channel 1 is transmitted via channels 1 and 2, channel 2 via 3 and 4 etc.
- The ADAT2 port transmits channels 5 to 8

Please refer to the diagram in chapter 31. Routing for record and playback is identical.

9.3 DVD-Playback (AC-3/DTS)

AC-3 / DTS

When using popular DVD software players like *WinDVD* and *PowerDVD*, their audio data stream can be sent to any AC-3/DTS capable receiver using the Fireface's SPDIF output. For this to work, the WDM SPDIF device of the Fireface 800 (MME is not supported by every software) has to be selected in 'Control Panel/Sounds and Multimedia/Audio'. Also check 'use preferred device only'.

The DVD software's audio properties now show the options 'SPDIF Out', 'Use SPDIF' or 'activate SPDIF output'. When selecting these, the software will transfer the non-decoded digital multichannel data stream to the Fireface.

This 'SPDIF' signal sounds like chopped noise at highest level. Therefore the Fireface 800 automatically activates the non-audio bit within the digital data stream, to prevent most SPDIF receivers from accepting the signal, and to prevent any attached equipment from being damaged.

Multichannel

PowerDVD and WinDVD can also operate as software decoder, sending a DVD's multichannel data stream directly to the analog outputs of the Fireface. All modes are supported, from 2 to 8 channels, at 16 bit resolution and up to 192 kHz sample rate. Up to 48 kHz, playback is also possible via the ADAT outputs of the Fireface.

For multichannel playback, a WDM streaming* device, or the correct** MME playback device of the Fireface 800 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 Fireface.

* The option *Interleaved* in the Settings dialog has to be activated.

** Certain MME playback devices of the Fireface support multichannel playback (Interleaved) with 2, 4, 6 and 8 channels:

- Fireface Analog (1+2): Up to 192 kHz, playback on the analog outputs 1-8
- Fireface ADAT1 (1+2): Up to 48 kHz, playback on channels 13-20
- Fireface ADAT2 (1+2): Up to 48 kHz, playback on channels 21-28

Thanks to TotalMix, the software's playback channels can be routed to any hardware outputs. Therefore a playback at 96 kHz using the analog outputs can also be done via both ADAT ports. This also removes a limitation in some software (like PowerDVD 5), which always uses channel 1 as first channel of a multichannel playback.

The channel assignment using PowerDVD is:

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

Note 1: Selecting the Fireface to be used as system playback device is against our recommendations on system optimization, 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 the selection after usage, or to disable any system sounds (tab Sounds, scheme 'No audio').

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

Note 3: In WinDVD 5 and 6, a channel-separated 5.1 playback using the Fireface requires a change in the registry. Start regedit, go to *HKEY_CURRENT_USER / Software / InterVideo / DVD5 (or 6) / 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!

9.4 Low Latency under MME (Buffer Size Adjustment)

Under 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 systems have become much more powerful, and with Windows 2000/XP latencies far lower can be used. SAWStudio and Sonar allowed to use such low settings from the start. Sequoia was updated in version 5.91, WaveLab in version 3.04.

In the Settings dialog the DMA buffer size is set with the buttons labeled **Buffer Size (Latency)**. This setting is valid for MME, ASIO and GSIF at the same time. Our test computers allow to use settings down to 48 samples without clicks. Please note that this setting only defines the buffer size of the hardware. The true and effective latency is configured within the MME application!



Attention: the DMA buffers must not be larger than the application's buffers. This case can happen unnoticed when using ASIO and MME at the same time (multi-client) and setting ASIO to 23 ms, while the buffers in the MME application are still set for a lower latency. Playback will be stuttering and audio will be distorted.

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

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

9.5 Notes on WDM

The driver offers two devices per stereo pair, like **Fireface Analog (1+2)** and **Fireface Analog (1+2) MME**.

Fireface Analog (1+2)

The devices having no MME at the end are **WDM Streaming Devices**. WDM Streaming is Microsoft's current driver and audio system, directly embedded into the operating system. WDM Streaming is nearly unusable for professional music purposes, as all data is processed by the so called Kernel Mixer, causing a latency of at least 30 ms. Additionally, WDM can perform sample rate conversions unnoticed, cause offsets between record and playback data, block channels unintentionally and much more. Therefore, for general operation, RME recommend not to use WDM devices.

WDM Streaming also replaces the former DirectSound. Synthesizers and Samplers, which achieved latencies below 10 ms using DirectSound, are forced to use WDM in Windows XP. And now operate at high latency. Meanwhile most of these programs support ASIO as low latency driver interface.

Several programs do not offer any direct device selection. Instead they use the *playback device* selected in Windows under *Control Panel/Sounds and Multimedia/Audio*. Such software often requires the special functions provided by WDM, and therefore will operate better when using a WDM device. Please note that selecting the Fireface to be used as system playback device is against our recommendations, as professional interfaces should not be disturbed by system events.

The program *Sonar* from Cakewalk is unique in many regards. Sonar uses the so called **WDM Kernel Streaming**, bypassing the WDM mixer, thus achieves a similar performance as ASIO and our MME driver (see below). The superior low latency operation that was originally expected from WDM Kernel Streaming has not become reality. This is not astonishing when analysing the driver model. Early reports about a sensational performance around 1 ms latency at up to 100% CPU load have been quickly revealed to be faulty drivers, operating at much higher latency than 1 ms.

Meanwhile Sonar supports ASIO too. So simply test and try for yourself which driver interface will work better for you on your system.

Information on multichannel WDM is found in chapter 9.6.

Fireface Analog (1+2) MME

The devices with the MME suffix bypass the Windows Kernel Mixer. This way, they achieve nearly the same performance as RME's previous (true) MME drivers. Also their behaviour (offsets, start/stop, multi-client) is identical. So when not using GSIF or ASIO, MME should be your preferred choice.

A small note on **ASIO** and **GSIF**: these drivers are completely independent from WDM Streaming. They work and perform the same as our previous 'just WDM' drivers.

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

ASIO-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. The 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.

Multi-Client and Multi-Channel using WDM

The WDM Streaming devices of our driver can operate as usual stereo devices, or as 8-channel devices. The option **Interleaved** in the Settings dialog determines the current mode.

Interleaved not active: The WDM devices operate as usual stereo devices. The multi-client operation works as described above with WDM, ASIO, MME and GSIF.

Interleaved active: The WDM devices can also be used as 8-channel devices. Unfortunately the Kernel Mixer, active with any WDM playback, then always occupies and blocks 8 channels at once, even when WaveLab or the Media Player perform just a stereo playback (2 channels). So:



If any stereo pair within an 8-channel group is used, the whole 8-channel group is blocked. As a result, no second stereo pair of this group can be used, neither with ASIO, MME nor GSIF.

The Fireface provides three 8-channel groups: the analog inputs 1 to 8 and the two ADAT ports.

Starting MME, ASIO or GSIF playback on any of the stereo pairs of an 8-channel group prior to starting a WDM playback will prevent the Kernel Mixer from opening the 8-channel device, as two of its channels are already in use. The Kernel Mixer then automatically reverts to open a stereo device for a stereo playback.

Software which is capable of using more than one MME device usually uses the (WDM-) MME devices. A surround playback via Media Player should be possible without the need of reconfiguration. That's why the mode **Interleaved** is activated in the Settings dialog by default.

9.7 Digital Recording

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

Taking this into account, RME have included three unique features in the Fireface 800: a comprehensive I/O signal status display showing sample frequency, lock and sync status in the Settings dialog, status LEDs for each input, and the protective *Check Input* function.

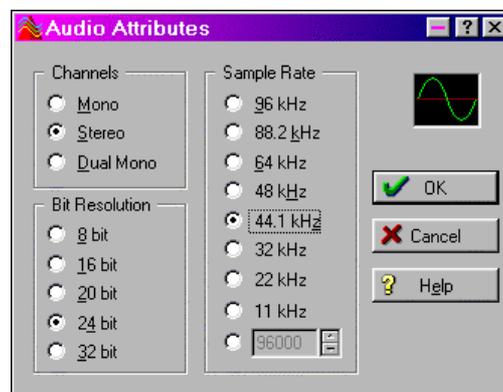
If a 48 kHz signal is fed to the input and the application is set to 44.1 kHz, *Check Input* stops the system from recording. This prevents faulty takes, which often go unnoticed until later on in the production. Such tracks appear to have the wrong playback rate - the audio quality as such is not affected.

The sample frequency shown in the Settings dialog (see chapter 8, screenshot Settings) is useful as a quick display of the current configuration (the board itself and all connected external equipment). If no sample frequency is recognized, it will read 'No Lock'.

This way, configuring any suitable audio application for digital recording is simple. After selecting the required input, Fireface 800 displays the current sample frequency. This parameter can then be changed in the application's audio attributes (or similar) dialogue.

The screenshot on 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 18).

Currently two solutions exist which enable an automated control of real-time monitoring. ZLM (Zero Latency Monitoring) allows monitoring in Punch I/O mode - with this the system behaves like a tape machine. This method has been implemented in all versions of Samplitude (by 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 in the Fireface 800.

The other solution is Steinberg's ASIO protocol with our ASIO 2.0 drivers and all ASIO 2.0 compatible programs (also 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.

9.8 Analog Recording

For recordings via the analog inputs the corresponding record device has to be chosen (Fireface Analog (x+x)). Apart from the three reference levels, the Fireface has no means to change the input level. This would make no sense for the digital inputs, but also for the analog inputs one can do without it. It doesn't matter if the Fireface is operated at a mixing desk or a multichannel mic preamp, in either case the level can be controlled directly at the source to match the Fireface's sensitivity perfectly.

The input sensitivity of the frontside analog inputs can be adjusted using their Gain pots to match any external source perfectly, see chapter 14.

10. Using more than one Fireface 800

The current driver supports up to three Fireface 800. 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 Firefaces is set to clock modus 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.
- TotalMix is part of the hardware of each Fireface. 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.
- 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 with ASIO, as the ASIO driver presents all units as one.
- When using all channels of more than one Fireface 800, a FireWire 800 interface is necessary. FireWire 400 will usually not suffice for operating more than one Fireface.
- When using only one Fireface 800, a FireWire 800 interface does not improve performance, and does not help to achieve lower latency. But connecting a hard drive to the Fireface (Hub functionality), FireWire 800 will immediately increase performance and reliability.

11. Operation under ASIO 2.0

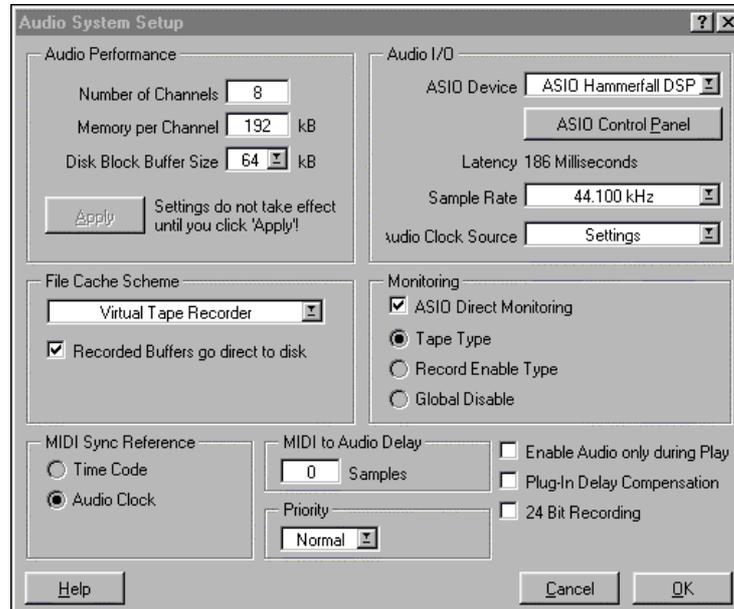
11.1 General

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

Start the ASIO software and select 'System' from the Audio menu. Select 'ASIO Fireface' as the audio I/O device. The 'ASIO system control' button opens the Fireface's Settings dialog (see chapter 8, Configuration).

Fireface 800 supports simultaneous record and playback of SPDIF audio data together with record and playback in ADAT format. Please note that the external SPDIF device has to be running in sync, otherwise recordings will be corrupted.

Fireface 800 supports 'ASIO Direct Monitoring' (ADM). Please note that currently Nuendo, Cubase and Logic do not support ADM completely error-free.



When the sample frequency is set to 88.2 or 96 kHz, the number of ASIO ADAT channels is reduced to 4 per ADAT port. At a sample rate of 176.4 or 192 kHz (Quad Speed Mode) the ADAT I/O is no longer available. Nevertheless it will send out a synchronized ADAT signal at a quarter of the sample rate. The ASIO driver corrects the number of channels when changing from Single to Double or Quad Speed. The channel routing of both ADAT ports is shown in chapter 31.

Some programs do not recognize the dynamic change of the number of ADAT channels, therefore do not operate correctly in Double and Quad Speed mode. A workaround is to use DDS. After having set the desired sample rate, the ASIO software recognizes the correct number of I/O channels directly when started.

11.2 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.

Unfortunately some newer UltraATA66 and UltraATA100 hard disk controllers (also Raid controllers) seem to 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').

The above note on PCI is not an error in this manual: so far all FireWire controllers are placed on the PCI bus. Therefore the same problems known from RMEs PCI cards might also show up with the Fireface 800.

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 Fireface 800 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!

12. Operation under GSIF (Gigasampler Interface)

Windows 2000/XP

The GSIF interface of the Fireface 800 allows direct operation with Gigastudio, with up to 28 channels, 96 kHz and 24 bit. The new GSIF 2.0 is also supported with both audio and MIDI.

Gigastudio requires a lot of a computer's CPU performance. Optimum performance is achieved with a stand-alone GSIF PC. However, when using the Fireface 800, 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 use audio channels exclusively on the Fireface 800. 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.

13. Stand-alone Operation

The Fireface 800 has an internal memory to permanently store all configuration data. These are:

Settings dialog

Sample rate, clock mode Master/Slave, configuration of the channels and the digital I/Os.

TotalMix

The complete mixer state.

The Fireface loads those settings directly after power-on. A simple, yet useful application is to store the correct clock mode, avoiding wrong clocking and noise disturbances in a complex setup, caused by wrong synchronization. Usually the unit will be configured by the Windows driver, so for the time between power-on of the computer up to the loading of the Windows driver its state might be wrong.

This total configuration feature in stand-alone operation, that is without any connected computer, turns the Fireface into lots of dedicated devices. Here are some ideas:

13.1 10-Channel AD/DA-Converter

When loading TotalMix' factory default 1 into the unit, the Fireface becomes a high quality 10-channel AD/DA-converter, which also provides a monitoring of all 8 DA-channels via channels 9/10 (Preset 2: also monitoring all 10 inputs). A small modification allows for a monitoring of all I/Os via the SPDIF I/O.

13.2 4-Channel Mic Preamp

Use TotalMix to route the 4 microphone inputs directly to the analog outputs. This turns the Fireface 800 into a 4-channel microphone preamp. The AD- and DA-conversion will cause a small delay of the signals of around 0.4 ms (at 192 kHz, see chapter 30.2). But this is not really relevant, as it is the same delay that would be caused by changing the microphone's position by about 14 centimeter (5.6 inches).

13.3 Monitor Mixer

TotalMix allows ANY configuration of all I/Os of the Fireface. For example, set up the device as monitor mixer for 10 analog signals, 16 digital via ADAT and 2 via SPDIF. Additionally, TotalMix lets you set up ANY submixes, so all existing outputs can be used for different and independent monitorings of the input signals. The perfect headphone monitor mixer!

13.4 Digital Format Converter

As TotalMix allows for any routing of the input signals, the Fireface 800 can be used as ADAT to SPDIF converter, ADAT to two ADAT splitter, and SPDIF to ADAT converter.

13.5 Analog/digital Routing Matrix

The Matrix in TotalMix enables you to route and link all inputs and outputs completely freely. All the above functionalities are even available simultaneously, can be mixed and combined in many ways. Simply said: the Fireface 800 is a perfect analog/digital Routing Matrix!

14. Analog Inputs

14.1 Line Rear

The Fireface has 8 balanced Line inputs as 1/4" TRS jacks on the back of the unit. The electronic input stage is built in a servo balanced design which handles unbalanced (mono jacks) and balanced (stereo jacks) correctly, automatically adjusting the level reference.



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

One of the main issues when working with an AD-converter is to maintain the full dynamic range within the best operating level. Therefore the Fireface 800 internally uses hi-quality electronic switches, which allow for a perfect adaptation of all rear inputs to the three most often used studio levels.

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

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

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

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

14.2 Microphone / Line Front

The balanced microphone inputs of the Fireface 800 offer an adjustable gain of 10 to 60 dB. The soft switching, hi-current Phantom power (48 Volt) provides a professional handling of condenser mics. The mic preamp's discreet Class-A front end guarantees a superior sound quality.

With the balanced Line input, which can be used alternatively or at the same time as the microphone, the Fireface 800 becomes even more flexible. 10 kOhm input impedance, stereo TRS jack and adjustable input sensitivity in a range of 50 dB – this all guarantees that the front-side Line inputs can be used perfectly with keyboards, sampler, active guitars and much more.

The Line inputs handle levels from -28 dBu up to +22 dBu. Two LEDs display a present signal (from -45 dBFS on) and warn against overload (-2 dBFS).

Channels 7/8 can be switched between Line rear, microphone, and Line/microphone simultaneously in the Settings dialog. The front Line input can be used simultaneously with the microphone input. This way, up to three different sound sources (Line rear, Line front, microphone) may be recorded at the same time on one channel.

14.3 Instrument

The instrument input of the Fireface 800 has been optimized especially for guitar and bass. A soft clipping function limits the level from -10 dBFS on, and offers tube-like distortion at full overload. The extra Drive stage adds even more distortion and also increased sustain. The Speaker Emulator gently shapes the sound for an optimal recording experience.

LIM

The distortion caused by the clipping function of the instrument input is audible. Depending on use and application the new harmonics can be nice or disturbing:

- The guitar's volume pot can blend from clean up to full distortion
- A rhythm guitar's sound is enhanced by a popular tube-like sound
- All dynamic peaks of a slapped bass guitar are removed without audible artifacts

- An accurately played clean acoustic guitar will sound crunchy
- Steady tones sound distorted (guitar / keyboards / bass)

Distortion can be completely avoided by adjusting the input level to never exceed -10 dBFS (so that the red LIM LED never lights up). This may sound unusual at first. But the AD-converters of the Fireface 800 are much better than the dynamic ratio of any instrument recording, so when intentionally not recording at fullest level, nothing is lost. As far as monitoring is concerned, TotalMix can compensate up to 6 dB, and the final recording usually is normalised within the software. In fact a headroom of 10 dB does not show any noticeable disadvantages.

Alternatively perfectly clean recordings at full level can be done via the Line inputs of the microphone channels. Active instruments can be connected directly and be level-adjusted perfectly with the Gain pots. Passive e-guitars require an additional impedance buffer. Most guitarists have one, but don't know that they do. Modern floor effect devices are active even in by-pass mode, and then operate as impedance buffer.

Drive

Drive is an additional clipper, for 25 dB more gain (=sustain) plus substantially higher distortion. It will simply blow you away. Also well suited for bass guitar.

Speaker Emulation

The basic idea of the Fireface 800 instrument input is not to alter the sound in a specific way, but to pre-condition the sound so that it gets much easier recorded and processed within the DAW application. This is accomplished by an optimization of the input/record signal via:

- light pre-clipping (see above)
- removing low frequency noise
- removing high-frequency noise
- a small bass and presence boost

All frequency corrections are part of the *Speaker Emulation* option. The name is originated by the fact that guitar cabinets typically show a big level attenuation in the high frequency range, making distortion sound less brilliant and harsh. After activating this option, a guitar completely distorted by LIM and Drive will sound excellent even when played directly into a mixing desk.

14.4 Notes on Input selection

Front and rear inputs of the Fireface 800 can be used alternatively or together. An inactive input will be muted. Front inputs are attenuated by more than 120 dB. The rear inputs will be attenuated by about 80 dB. Therefore in specific cases the rear inputs can bleed through into the front inputs. When using the front inputs, make sure that no high level signals are present at the corresponding rear inputs.

15. Analog Outputs

15.1 Line

The 8 short circuit protected, low impedance line outputs are available as 1/4" TRS jacks on the back of the unit. The electronic output stage is built in a servo balanced design which handles unbalanced (mono jacks) and balanced (stereo jacks) correctly.

To maintain an optimum level for devices connected to the analog outputs, the Fireface 800 internally uses hi-quality electronic switches, which allow for a perfect adaptation of all outputs to the three most often used studio levels.

As with the analog inputs, the analog output levels are defined to maintain a problem-free operation with most other devices. The headroom of the Fireface 800 lies between 9 and 15 dB, according to the chosen reference level:

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

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

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

15.2 Headphones

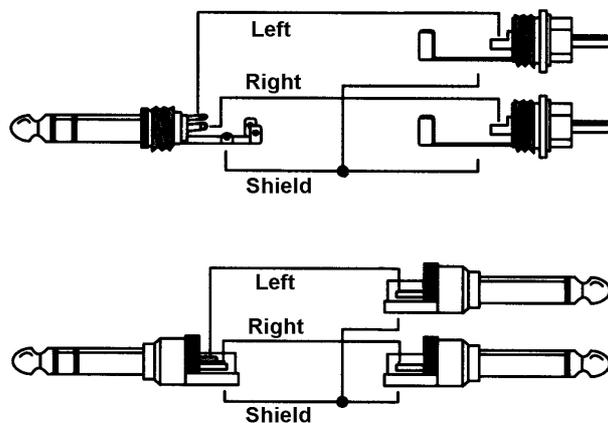
Channels 9/10 of the Fireface are available on the front via one 1/4" unbalanced TRS jack (stereo output). These channels use the same converters as the other Line outputs, therefore offer the same technical data (119 dBA SNR!).

Instead of using internal electronic switches, their output level is changed step-less with the VOL pot. These outputs are special low impedance types, ready to be used with headphones. But they can also be used as high-quality (yet unbalanced) Line outputs.

Like all other outputs, channels 9/10 can also be controlled by TotalMix regarding level and monitoring of any input or playback channels (submix, like factory presets 1 and 2).

In case the output should operate as line output, an adapter TRS plug to RCA phono plugs, or TRS plug to TS plugs is required.

The pin assignment follows international standards. The left channel is connected to the tip, the right channel to the ring of the TRS jack/plug.



16. Digital Connections

16.1 ADAT

The ADAT optical inputs of the Fireface 800 are fully compatible with all ADAT optical outputs. RME's unsurpassed Bitclock PLL prevents clicks and drop outs even in extreme varipitch operation, and guarantees a fast and low jitter lock to the digital input signal. A usual TOSLINK cable is sufficient for connection. More information on Double Speed (S/MUX) can be found in chapter 30.1.

ADAT1 IN

Interface for the first or only device sending an ADAT signal to the Fireface 800. Carries the channels 1 to 8. When receiving a Double Speed signal, this input carries the channels 1 to 4.

ADAT2 IN

Interface for the second device sending an ADAT signal to the Fireface 800. Carries the channels 9 to 16. When receiving a Double Speed signal, this input carries the channels 5 to 8. Can also be used as SPDIF optical input.

ADAT1 Out

Interface for the first or only device receiving an ADAT signal from the Fireface 800. Transmits channels 1 to 8. When sending a Double Speed signal, this port carries channels 1 to 4.

ADAT2 Out

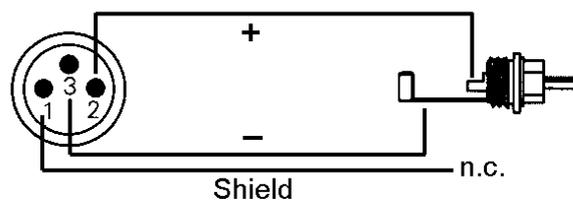
Interface for the second device receiving an ADAT signal from the Fireface 800. Transmits channels 9 to 16. When sending a Double Speed signal, this port carries channels 5 to 8. Can also be used as SPDIF optical output.

16.2 SPDIF

The SPDIF input is configured in the Settings dialog, available by a click on the fire symbol in the Task Bar's system tray. The Fireface 800 accepts all commonly used digital sources as well as SPDIF and AES/EBU. Channel status and copy protection are ignored.

In SPDIF mode, identical signals are available at both the optical and the coaxial output. An obvious use for this would be to connect two devices, i.e. using the Fireface 800 as a splitter (distribution 1 on 2).

To receive signals in AES/EBU format, an adapter cable is required. Pins 2 and 3 of a female XLR plug are connected individually to the two pins of a phono plug. The cable shielding is only connected to pin 1 of the XLR - not to the phono plug.



The ground-free design, with transformers for coaxial digital inputs and outputs, offers a trouble-free connection of all devices along with perfect hum rejection and full AES/EBU compatibility.

Special Characteristics of the SPDIF Output

Apart from the audio data itself, digital signals in SPDIF or AES/EBU format have a header containing channel status information. False channel status is a common cause of malfunction. The Fireface 800 ignores the received header and creates a totally new one for the output signal.



Note that in record or monitor modes, set emphasis bits will disappear. Recordings originally done with emphasis should always be played back with the emphasis bit set!

This can be done by selecting the *Emphasis* switch in the Settings dialog (SPDIF Out). This setting is updated immediately, even during playback. The Fireface's new output header is optimized for largest compatibility with other digital devices:

- 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz depending on the current sample rate
- Audio use, Non-Audio
- No Copyright, Copy Permitted
- Format Consumer or Professional
- Category General, Generation not indicated
- 2-channel, No Emphasis or 50/15 μ s
- Aux bits Audio Use

Professional AES/EBU equipment can be connected to the Fireface 800 thanks to the transformer-balanced coaxial outputs, and the 'Professional' format option with doubled output voltage. Output cables should have the same pinout as those used for input (see above), but with a male XLR plug instead of a female one.



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

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

16.3 MIDI

Fireface 800 offers one MIDI I/O via 5-pin DIN jacks. The MIDI ports are added to the system by the driver. Using MIDI capable software, these ports can be accessed under the name *Fireface Midi*. Using more than one Fireface, the operating system adds a consecutive number to the port name, like *Fireface MIDI (1)* etc.

The MIDI inputs can not operate multi-client, so a MIDI input signal can not be distributed to several programs at the same time. Such a functionality can be achieved with third party tools.

17. Word Clock

17.1 Word Clock Input and Output

SteadyClock guarantees an excellent performance in all clock modes. Based on the highly efficient jitter suppression, the Fireface refreshes and cleans up any clock signal, and provides it as reference clock at the BNC output (see section 24).

Input

The Fireface'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 Fireface 800 automatically adapts to it. As soon as a valid signal is detected, the WC LED is lit, and the Settings dialog shows either Lock or Sync (see chapter 30.1).

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 Fireface's word clock input is shipped as high impedance type (not terminated). A push switch allows to activate internal termination (75 Ohms). The switch is found on the back beside the word clock input socket. Use a small pencil or similar and carefully push the blue switch so that it snaps into its lock position. The yellow LED will be lit when termination is active. Another push will release it again and de-activate the termination.



Output

The word clock output of the Fireface 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. In all other cases the provided frequency is identical to the one present at the currently chosen clock input. When the current clock signal fails, the Fireface 800 switches to Master mode and adjusts itself to the next, best matching frequency (44.1 kHz, 48 kHz etc.).

Selecting *Single Speed* in the Settings dialog 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.

The received word clock signal can be distributed to other devices by using the word clock output. With this the usual T-adaptor can be avoided, and the Fireface 800 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 also with most critical word clock signals

Thanks to a low impedance, but short circuit proof output, the Fireface delivers 4 Vpp to 75 Ohms. For wrong termination with 2 x 75 Ohms (37.5 Ohms), there are still 3.3 Vpp at the output.

17.2 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 and ADAT 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! If the Fireface's clock mode is set to 'Master', all other devices must be set to 'Slave'.

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 typically 15 times higher as when using a quartz based clock.

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

The actual end of these problems is offered by the **SteadyClock** technology of the Fireface 800. Combining the advantages of modern and fastest digital technology with analog filter techniques, re-gaining a low jitter clock signal of 11 MHz from a slow word clock of 44.1 kHz is no problem anymore. Additionally, jitter on the input signal is highly rejected, so that even in real world usage the re-gained clock signal is of highest quality.

17.3 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 2×75 Ohms, and the user is forced to buy a special word clock distributor. Note that such a device is generally recommended for bigger studios.

Also, 75 Ohm cable is hard to find these days. 50 Ohm cable is standard – and will work as long as the termination resistors are 75 Ohm.

The Fireface's word clock input can be high-impedance or terminated internally, ensuring maximum flexibility. If termination is necessary (e.g. because the Fireface is the last device in the chain), push the switch at the back beside the BNC socket (see chapter 17.1).

In case the Fireface 800 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 Fireface 800, we recommend not to pass the input signal via T-adapter, but to use the Fireface'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.

17.4 Operation

The green Lock LED on the front (DIGITAL STATE) will light up as soon as a word clock signal is detected. To change to word clock as clock source, active clock mode *AutoSync* and switch *Pref. Sync Ref* to *Word Clock*. The status display *AutoSync Ref* changes to *Word* as soon as a valid signal is present at the BNC jack. This message has the same meaning as the green *Lock* LED, but appears on the monitor, i.e. the user can check immediately whether a valid word clock signal is present and is currently being used.

AutoSync Ref also displays the frequency (*Freq.*) of the reference signal, here the frequency of the current word clock signal, measured by the hardware.

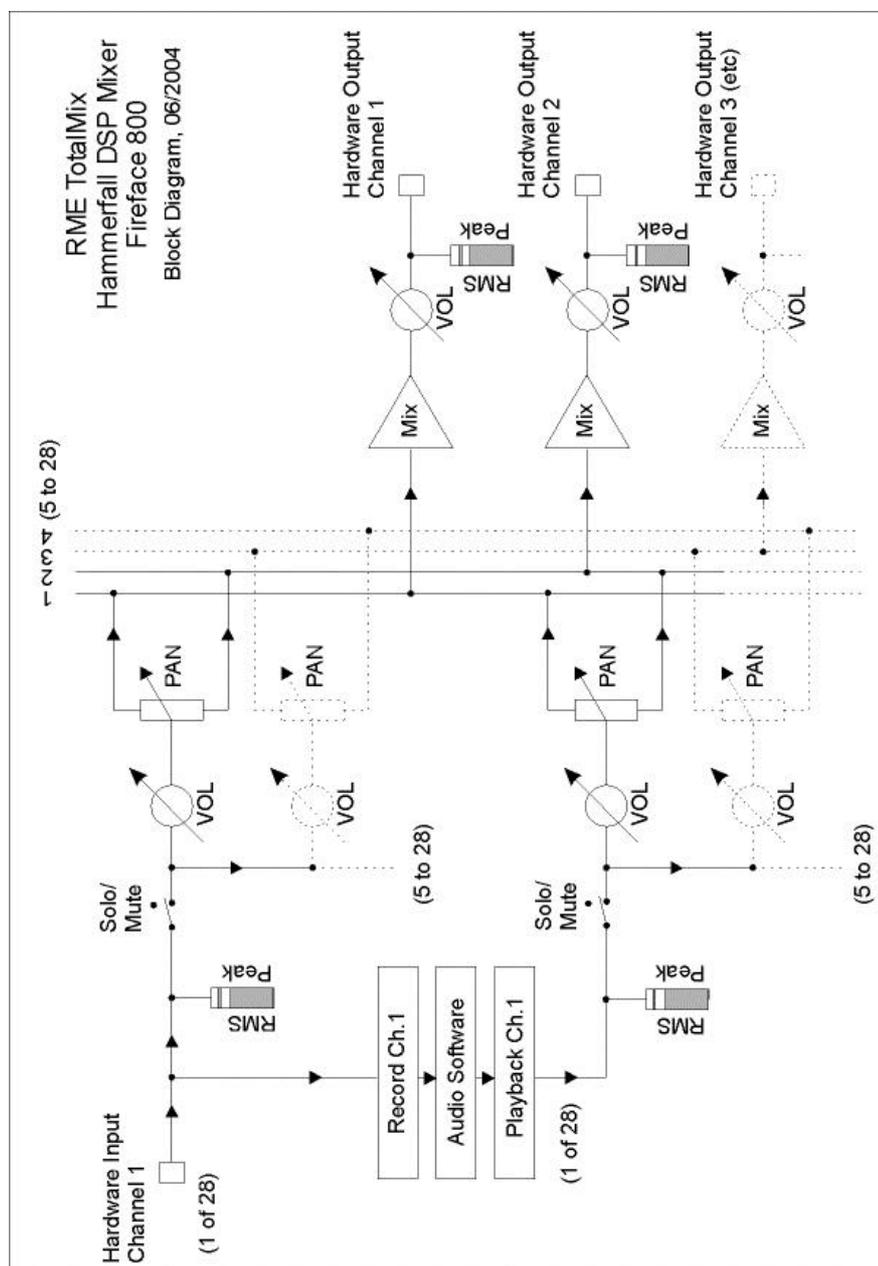
18. TotalMix: Routing and Monitoring

The Fireface 800 includes a powerful digital real-time mixer. RME's unique TotalMix technology allows for nearly unlimited mixing and routing with all inputs and playback channels simultaneously.

Here are some typical applications for TotalMix:

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

The block diagram of the TotalMix mixer of the Fireface 800 shows that the record signal always stays un-altered, but can be passed on as often as desired, even with different levels. The level meter of inputs and playback channels are connected pre-fader (due to the enormous routing capabilities). The level meters of the hardware's outputs are connected post-fader.



18.1 Elements of the Surface

The visible design of the TotalMix mixer is mainly determined by the architecture of the Fireface 800:

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

Every single channel has various elements:

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

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

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

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

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

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



18.2 Tour de TotalMix

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

We will now create a submix on analog outputs 7/8. Please start a multitrack playback. In playback channel 1 (labeled 'Out 1'), click onto the routing window below the label. A list pops up, showing a checkmark in front of 'AN 1+2'. Click onto 'AN 7+8'. The list disappears, the routing window no longer shows 'AN1+2', but 'AN 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 also 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.

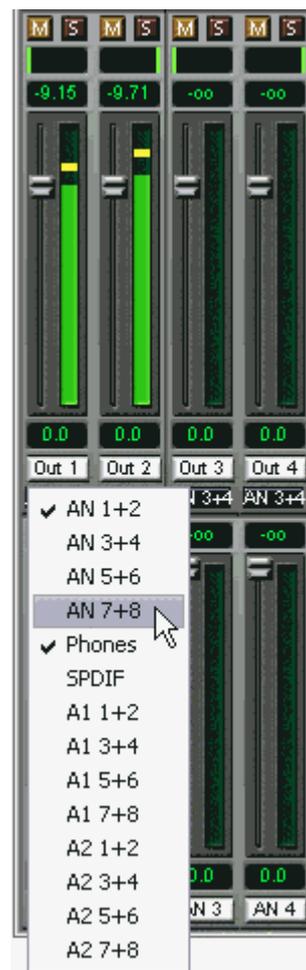
Often signals are stereo, i. e. a pair of two channels. It is therefore helpful to be able to make the routing settings for two channels at once. Press the Ctrl-key and click into the routing window of 'Out 3' with the key pressed. The routing list pops up with a checkmark at 'AN 3+4'. Click onto 'AN 7+8'. Now, Out 4 has already been set to 'AN 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 window. You'll now see two checkmarks, one at 'AN 3+4', the other one at 'AN 7+8'. Click onto 'SPDIF'. The window disappears, fader and panpot jump to their initial values, the signal can now be routed to these channels. You can continue, until all entries have got a checkmark, i. e. you can send the signal to all outputs simultaneously.

You will certainly have noticed that the signal at the outputs 7/8 did not change, while you were routing channel 4 to other outputs and setting different gain values. With all analog and most digital mixing desks, the fader setting would affect the level for every routed bus - not so for TotalMix. TotalMix allows for setting all fader values individually. Therefore the faders and the panpots jump to the appropriate setting as soon as another routing is chosen.

The checkmarks are un-checked by moving the fader to m.a. This setting deactivates the routing...why route if there is no level? Click onto 'AN 7+8' in the routing window, pull the fader down, open the routing window again - the checkmark is gone.



18.3 Submix View

Such a wide range of possibilities make it difficult to maintain the overview. Because practically all hardware outputs can be used for different submixes, as shown (up to 14 completely independent stereo submixes, 7 4-channel submixes etc.). And when opening the routing windows you might see an army of checkmarks, but you don't get an overview, i.e., how the signals come together and where. This problem is removed by the view mode 'Submix'. In this mode, all routing windows jump to the routing pair just being selected. You can then see immediately, which channels, which fader and pan settings make a submix (for example 'AN 7+8').

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.

18.4 Mute and Solo

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

The same holds true for the Solo and the *Solo Master* buttons. Solo is working as a solo-in-place. As soon as one Solo button is pressed, all other Mute buttons are activated and light up.

18.5 Hotkeys

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

- Clicking anywhere into the fader area with the Ctrl-key pressed, sets the fader to 0 dB, -6 dB for the hardware outputs.
- Clicking anywhere into the pan area with the Ctrl-key pressed, sets the panorama to <C> meaning 'Center'.
- Clicking a Preset button while holding down Ctrl, the original (factory) preset will be loaded.
- Using Ctrl and any number between 1 and 8 (not on the numeric keypad!) will load the corresponding factory default preset (Alt loads the user preset)
- Clicking the Card 2 button while holding down Ctrl opens a second TotalMix window.

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

At the same time, TotalMix also supports combinations of these keys. If you press Ctrl and Alt at the same time, clicking with the mouse makes the faders jump to 0 dB pairwise, and they can be set pairwise by Shift-Alt in fine-mode.

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

Using the hotkeys I, O and P the complete row each of Input, Playback and Output channels can be toggled between visible and invisible. Hotkey S switches Submix view on/off. Those four hotkeys have the same functionality as the buttons in the View section of the Quick Access Panel. The Level Meter Setup dialog can be opened via F2 (as in DIGICheck).

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 19), 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 18.9):

- 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

18.6 The Quick Access Panel

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

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

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

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

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

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

Up to three Firefaces can be used simultaneously. The **Card** buttons switch between the cards. Holding down Ctrl while clicking on button Card2 will open a second window, instead of replacing the current window content.

The number of ADAT channels is reduced automatically when entering Double Speed mode (88.2 and 96 kHz). The display is adjusted accordingly, and all fader settings remain stored (even the invisible ones).



18.7 Presets

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

This method offers two major advantages:

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

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 (Alt loads the user presets)

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

Preset1.mix

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

Details: All inputs maximum attenuation (m.a.). All playback channels 0 dB, routed to the same output. All outputs 0 dB, Phones -6 dB. Submix of all inputs and playbacks to channel 9/10 (Phones). All channels prepared for all routings to left/right panning. Level display RMS +3 dB.

|

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

Preset2.mix

Description: All channels routed 1:1, input and playback monitoring via Phones. As Preset 1, plus submix of all inputs (0 dB) to channels 9/10 (Phones).

Preset3.mix

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

Preset4.mix

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

Preset5.mix

Description: Playback monitoring to Phones. As Preset 1, but all outputs except channels 9/10 (Phones) set to maximum attenuation.

Preset6.mix

Description: All channels routed 1:1, monitoring of all playback channels via Phones and SPDIF. As Preset 1, plus submix of all playbacks to SPDIF.

Preset7.mix

Description: Monitoring of all playback channels via Phones and of all record/playback channels via SPDIF. As Preset 6, plus submix of all inputs to SPDIF.

Preset8.mix

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

18.8 Menu Options

Always on Top: When active (checked) the window of TotalMix will always be on top of the Windows desktop.

Note: This function may result in problems with windows containing help text, as the TotalMix window will even be on top of those windows, so the help text isn't readable.

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

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

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

Link Faders: Selecting this option all faders will be treated as stereo pairs. Equals a held down Alt-key. Hotkey L.

Level Meter Setup: Configuration of the Level Meters. Hotkey F2. See chapter 18.9.

Flash current mixer state: A click on this entry stores all current mixer settings into the flash memory of the Fireface. See also chapter 13, Stand-alone Operation.

18.9 Level Meter

The Fireface 800 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.

The level meters integrated in TotalMix - considering their size - cannot be compared with DIGICheck (chapter 21). Nevertheless they already include many useful functions.

Peak and RMS is displayed for every channel. 'Level Meter Setup' (Menu Options or F2) or direct keyboard entry (*hotkeys*) makes various options available:

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

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

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

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

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



19. The Matrix

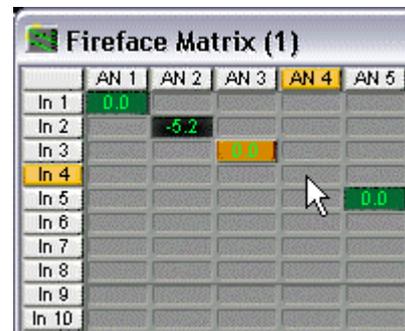
The mixer window of TotalMix looks and operates similar to mixing desks, as it is based on a conventional stereo design. The matrix display presents a different method of assigning and routing channels, based on a single channel or monaural design. The matrix view of the Fireface 800 looks and works like a conventional patchbay, adding functionality way beyond comparable hardware and software solutions. 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.

19.1 Elements of the Surface

The visible design of the TotalMix Matrix is mainly determined by the architecture of the Fireface 800:

- Horizontal labels: All hardware outputs
- Vertical labels: All hardware inputs. Below are all playback channels (software playback)
- Green 0.0 dB field. Standard 1:1 routing
- Black gain field. Shows the current gain value as dB
- Orange gain field. This routing is muted.



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

19.2 Operation

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

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

Note the difference between the left side, representing the inputs and software playback sources, and the upper side, representing the hardware outputs. If you move a fader in row 1 or 2 in TotalMix, only the specific levels (max. 2) of this routing will change in the Matrix. But moving a fader in row 3 will make all vertically activated levels move at once (for example 9/10, phones output).

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

19.3 Advantages of the Matrix

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

Example 1: You want TotalMix to route all software outputs to all corresponding hardware outputs, and have a submix of all inputs and software outputs on the analog output (equals factory preset 2). Setting up such a submix is easy. But how to check at a later time, that all settings are still exactly the way you wanted them to be?

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

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

20. TotalMix Super-Features

20.1 ASIO Direct Monitoring

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

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

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

20.2 Selection and Group-based Operation

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

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

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

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

20.3 Copy Routings to other Channels

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

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

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

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

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

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

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

20.4 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!

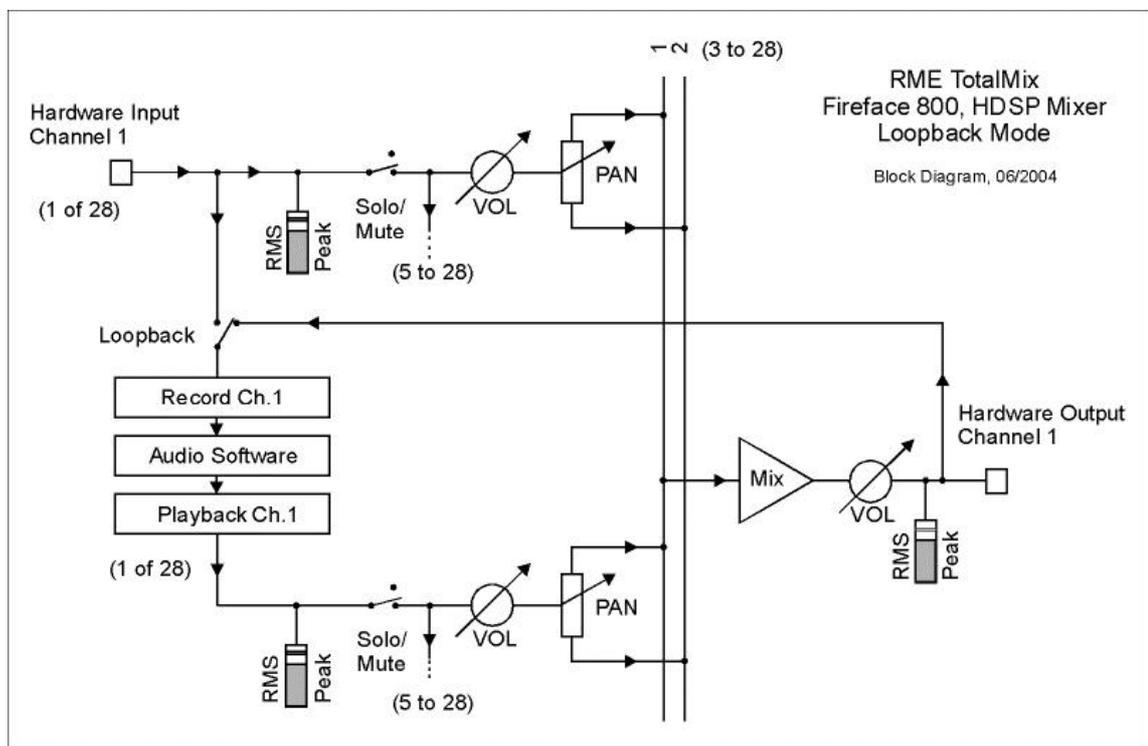
20.5 Recording a Subgroup (Loopback)

TotalMix supports a routing of the subgroup outputs (=hardware outputs) 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.

In this mode, 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 28 hardware outputs can be routed to the record software, and none of these hardware inputs get lost, TotalMix offers a 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 the software or 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 it via Ctrl-mouse for recording.

21. DIGICheck

The DIGICheck software is a unique utility developed for testing, measuring and analysing digital audio streams. Although the DIGICheck software is fairly self-explanatory, it still includes a comprehensive online help. DIGICheck 4.2 operates as multi-client ASIO host, and can therefore be used in parallel to any software, be it WDM, MME, ASIO or GSIF, both inputs and even outputs (!). The following is a short summary of the available functions:

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

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

Note: The dedicated Meter Bridge known from RME's HDSP series, a professional level display nearly without CPU load, will be added soon to DIGICheck. It can then be used with the Fireface 800 too.

22. Hotline – Troubleshooting

22.1 General

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

The input signal cannot be monitored in real-time

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

The 8 ADAT channels don't seem to work

- The optical output ADAT2 has been switched to SPDIF. As can be seen in the block diagram, all channels and their assignments still exist, but the optical transmitter has been disconnected from ADAT2 and is now fed from the SPDIF output (channels 11/12). The ADAT2 playback devices are still usable by routing and mixing them in TotalMix to other outputs.

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 Fireface 800 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.

22.2 Installation

After the driver installation the Fireface 800 is found in the Device Manager (>Settings/Control Panel/System<), category 'Sound, Video and Gamecontroller'. A double click on 'RME Fireface 800' starts the properties dialog. The well known entry 'Resources' does not exist, because the Fireface is an external device connected to the FireWire Controller. The latter can be used to check interrupt and memory range.

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



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!

Possible causes for a Fireface not found automatically:

- The FireWire port is not active in the system (drivers of the FireWire PCI or CardBus card have not been installed)
- The FireWire cable is not, or not correctly inserted into the socket
- No power. After switching the Fireface on, at least the red Host error LED has to be lit.
- The PCI FireWire card is defective

A report about incompatible FireWire 800 controllers is found in the Tech Info *FireWire 800 Hardware – Compatibility Problems*.

23. Noise level in DS / QS Mode

The outstanding signal to noise ratio of the Fireface's AD-converters can be verified even without expensive test equipment, by using our famous *DIGICheck* tool or the record level meter of Steinberg's *WaveLab*. When activating the DS and QS mode, the displayed noise level will rise from -109 dB to -104 dB at 96 kHz, and -82 dB at 192 kHz. This is not a failure. This kind of measurement measures the noise of the whole frequency range, at 96 kHz from 0 Hz to 48 kHz (RMS unweighted), at 192 kHz even from 0 Hz to 96 kHz.

When limiting the measured area to 22 kHz (audio bandpass, weighted) the value would be -110 dB again. This can be verified even with RME's DIGICheck. Although a dBA weighted value does not include such a strong bandwidth limitation as audio bandpass does, the displayed value of -108 dB is nearly identical to the one at 48 kHz.

Subframe	MSB	Audio Data	LSB	AUX	CUV	RMS [dB+3]	RMS [dBA+3]	DC [dB]
1 - Left	xxxx	xxxxxxxx	xxxxxxxx	xxxx	00	-81.9	-108.5	LOW
2 - Right	xxxx	xxxxxxxx	xxxxxxxx	xxxx	00	-82.1	-108.9	LOW
Bits	4	8	12	16	20	24		

Press F1 for help. SR 192kHz

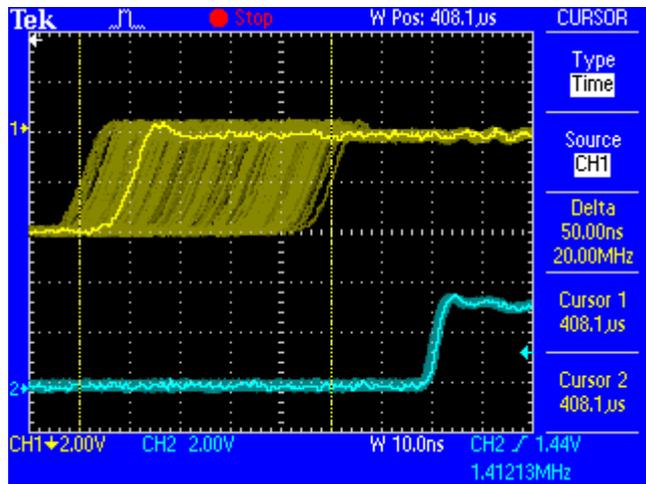
The reason for this behaviour is the noise shaping technology of the analog to digital converters. They move all noise and distortion to the in-audible higher frequency range, above 24 kHz. That's how they achieve their outstanding performance and sonic clarity. Therefore the noise is slightly increased in the ultrasound area. High-frequent noise has a high energy. Add the doubled (quadrupled) bandwidth, and a wideband measurement will show a significant drop in SNR, while the human ear will notice absolutely no change in the audible noise floor.

24. SteadyClock

The SteadyClock technology of the Fireface 800 guarantees an excellent performance in all clock modes. Thanks to a highly efficient jitter suppression, the AD- and DA-conversion always operates on highest sonic level, being completely independent from the quality of the incoming clock signal.

SteadyClock has been originally developed to gain a stable and clean clock from the heavily jittery MADI data signal (the embedded MADI clock suffers from about 80 ns jitter). Using the ADI-2's input signals SPDIF and ADAT, you'll most probably never experience such high jitter values. But SteadyClock is not only ready for them, it would handle them just on the fly.

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



The screenshot shows an extremely jittery SPDIF signal of about 50 ns jitter (top graph, yellow). SteadyClock turns this signal into a clock with less than 2 ns jitter (lower graph, blue). The signal processed by SteadyClock is of course not only used internally, but also used to clock the digital outputs. Therefore the refreshed and jitter-cleaned signal can be used as reference clock without hesitation.

25. Accessories

RME offers several optional components for the Fireface 800:

Part Number	Description
36001	Firewire cable IEE1394a 6M/6M, 1 m (3.3 ft)
36002	Firewire cable IEE1394a 6M/6M, 3 m (9.9 ft)
36005	Firewire cable IEE1394a 6M/6M, 5 m (16.4 ft)

FireWire cable for any FireWire 400 device, both sides 6-pin male. Cable longer than 16 ft is not specified for FireWire.

36003	Optical cable, TOSLINK, 0.5 m (1.6 ft)
36004	Optical cable, TOSLINK, 1 m (3.3 ft)
36006	Optical cable, TOSLINK, 2 m (6.6 ft)
36007	Optical cable, TOSLINK, 3 m (9.9 ft)
36008	Optical cable, TOSLINK, 5 m (16.4 ft)
36009	Optical cable, TOSLINK, 10 m (33 ft)

Standard lightpipe with TOSLINK connectors, RME approved quality.

TCOFF Time Code Option Fireface

Time Code Option to be inserted in a rear slot, adding LTC and Video synchronization inputs to the Fireface.

26. TECH INFO

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

Synchronization II (DIGI96 series)

Digital audio synchronization - technical background and pitfalls.

Installation problems

Problem descriptions and solutions.

Information on driver updates

Lists all changes in the drivers.

Configuring Logic, Samplitude, Cubase, Cakewalk, Sonar and SAWPlus32

Step by step instructions for use with RME cards.

DIGICheck: Analysis, tests and measurements with the DIGI96 series

A description of DIGICheck, including technical basics.

ADI-8 Inside

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

HDSP System: Notebook Basics - Notebook Hardware

HDSP System: Notebook Basics - The Audio Notebook in Practice

HDSP System: Notebook Basics - Background Knowledge and Tuning

HDSP System: Notebook Tests - Compatibility and Performance

Many background information on laptops. Tests of notebooks

HDSP System: TotalMix - Hardware and Technology

HDSP System: TotalMix - Software, features, operation

The digital mixer of the Hammerfall DSP in theory and practise

FireWire 800 – Compatibility Problems

27. Warranty

Each individual Fireface 800 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 Fireface 800. The general terms of business drawn up by Synthax Audio AG apply at all times.

28. Appendix

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

<http://www.rme-audio.com>

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

Manufacturer:

IMM Elektronik GmbH, Leipziger Strasse 32, D-09648 Mittweida

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Current driver version: W2k/XP: 1.0

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29. Technical Specifications

29.1 Analog

AD, Line In 1-8, rear

- Resolution AD: 24 bit
- Signal to Noise ratio (SNR): 110 dB RMS unweighted, 113 dBA
- Frequency response @ 44,1 kHz, -0.1 dB: 5 Hz – 21.5 kHz
- Frequency response @ 96 kHz, -0.5 dB: 5 Hz – 45.3 kHz
- Frequency response @ 192 kHz, -1 dB: 5 Hz - 74 kHz
- THD: < -110 dB, < 0.00032 %
- THD+N: < -104 dB, < 0.00063 %
- Channel separation: > 110 dB
- Maximum input level: +19 dBu
- Input: 6.3 mm TRS jack, electronically balanced
- Input impedance: 10 kOhm
- Input sensitivity switchable to Lo Gain, +4 dBu, -10 dBV
- Input level for 0 dBFS @ Lo Gain: +19 dBu
- Input level for 0 dBFS @ +4 dBu: +13 dBu
- Input level for 0 dBFS @ -10 dBV: +2 dBV
- Mute: 80 dB

Line In 7-10, Front

- as AD, but:
- Gain range (via pot): 50 dB
- Maximum input level, Gain 10 dB: +22 dBu
- Maximum input level, Gain 60 dB: -28 dBu
- CLIP LED: -2 dBFS
- SIG LED: -45 dBFS
- Mute: > 130 dB

Microphone

- as AD, but:
- Input: XLR, electronically balanced
- Input impedance: 2 kOhm
- Gain range (via pot): 50 dB
- Maximum input level, Gain 10 dB: +11 dBu
- Maximum input level, Gain 60 dB: -39 dBu
- CLIP LED: -2 dBFS
- SIG LED: -45 dBFS
- Mute: > 130 dB

Instrument

- as AD, but:
- Input: 6.3 mm TS jack, unbalanced
- Input impedance: 470 kOhm
- Signal to Noise ratio (SNR): 108 dB RMS unweighted, 111 dBA
- Gain range (via pot): 52 dB
- Drive Gain: 23 dB
- Maximum input level, Gain 1, -10 dBFS: +10 dBu
- Maximum input level, Gain 9, -10 dBFS: -42 dBu
- LIM LED: -10 dBFS
- SIG LED: -40 dBFS
- Mute: > 120 dB

DA, Line Out 1-8, rear

- Resolution: 24 bit
- Dynamic range (DR): 116 dB, 119 dBA @ 44.1 kHz (unmuted)
- Frequency response @ 44,1 kHz, -0,1 dB: 1 Hz – 21.1 kHz
- Frequency response @ 96 kHz, -0.5 dB: 1 Hz – 43.5 kHz
- Frequency response @ 192 kHz, -1 dB: 1 Hz - 70 kHz
- THD: -103 dB, < 0.0007 %
- THD+N: -100 dB, < 0.0001 %
- Channel separation: > 110 dB
- Maximum output level: +19 dBu
- Output: 6.3 mm TRS jack, servo-balanced
- Output impedance: 75 Ohm
- Output level switchable Hi Gain, +4 dBu, -10 dBV
- Output level at 0 dBFS @ Hi Gain: +19 dBu
- Output level at 0 dBFS @ +4 dBu: +13 dBu
- Output level at 0 dBFS @ -10 dBV: +2 dBV

DA - Stereo Monitor Output (Phones)

- as DA, but:
- Output: 6.3 mm TRS jack, unbalanced (stereo)
- Maximum output level at 0 dBFS: +17 dBu
- Output impedance: 30 Ohm

29.2 Digital

- Clocks: Internal, ADAT In, SPDIF In
- Low Jitter Design: < 1 ns in PLL mode, all inputs
- Internal clock: 800 ps Jitter, Random Spread Spectrum
- Jitter suppression of external clocks: about 30 dB (2.4 kHz)
- Effective clock jitter influence on AD and DA conversion: near zero
- PLL ensures zero dropout, even at more than 100 ns jitter
- Digital Bitclock PLL for trouble-free varispeed ADAT operation
- Internal sample rates: 32 / 44.1 / 48 / 64 / 88.2 / 96 / 128 / 176.4 / 192 kHz

29.3 Digital Inputs

AES/EBU - SPDIF

- 1 x RCA, transformer-balanced, galvanically isolated, according to AES3-1992
- High-sensitivity input stage (< 0.3 Vss)
- SPDIF compatible (IEC 60958)
- Accepts Consumer and Professional format, copy protection will be ignored
- Lock Range: 27 kHz – 200 kHz
- Jitter when synced to input signal: < 1 ns
- Jitter suppression: about 30 dB (2.4 kHz)

ADAT Optical

- 2 x TOSLINK
- Standard: 16 channels 24 bit, up to 48 kHz
- Sample Split (S/MUX): 2 x 8 channels 24 bit / 48 kHz, equalling 8 channels 24 bit 96 kHz
- Bitclock PLL ensures perfect synchronisation even in varispeed operation
- Lock Range: 31.5 kHz – 50 kHz
- Jitter when synced to input signal: < 1 ns
- Jitter suppression: about 30 dB (2.4 kHz)

Word Clock

- BNC, not terminated (10 kOhm)
- Switch for internal termination 75 Ohm
- Automatic Double/Quad Speed detection and internal conversion to Single Speed
- SteadyClock guarantees super low jitter synchronization even in varispeed operation
- Transformer coupled, galvanically isolated input
- Not affected 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 Vss
- Lock Range: 27 kHz – 200 kHz
- Jitter when synced to input signal: < 1 ns
- Jitter suppression: about 30 dB (2.4 kHz)

29.4 Digital Outputs

AES/EBU - SPDIF

- 1 x RCA, transformer-balanced, galvanically isolated, according to AES3-1992
- Output level Professional 2.5 Vpp, Consumer 1.2 Vpp
- Format Professional according to AES3-1992 Amendment 4
- Format Consumer (SPDIF) according to IEC 60958
- Single Wire mode, sample rate 28 kHz up to 200 kHz

ADAT

- 2 x TOSLINK
- Standard: 16 channels 24 bit, up to 48 kHz
- Sample Split (S/MUX): 2 x 8 channels 24 bit / 48 kHz, equalling 8 channels 24 bit 96 kHz
- In Quad Speed mode output of Single Speed sync frame

Word Clock

- BNC
- Max. output voltage: 5 Vpp
- Output voltage @ 75 Ohm termination: 4.0 Vpp
- Output impedance: 10 Ohm
- Frequency range: 27 kHz – 200 kHz

29.5 MIDI

- 1 x MIDI I/O via 5-pin DIN jacks
- Galvanically isolated by optocoupled input
- Hi-speed mode: Jitter and response time typically below 1 ms
- Separate 128 byte FIFOs for input and output

29.6 Transfer Modes: Resolution / Bits per Sample

ASIO

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

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

MME

- 16 bit 2 byte (stereo 4 bytes)
- 16 bit 4 byte MSB (stereo 8 bytes)
- 24 bit 4 byte MSB (stereo 8 bytes)
- 32 bit 4 byte (stereo 8 bytes)

MME interleaved supports 2-channel, 4-channel, 6-channel and 8-channel playback in the above formats. The modes 4/6/8-channel playback are limited to the devices

Fireface Analog (1+2): Up to 192 kHz, playback on the analog outputs 1-8

Fireface ADAT1 (1+2): Up to 48 kHz, playback on channels 13-20

Fireface ADAT2 (1+2): Up to 48 kHz, playback on channels 21-28

30. Technical Background

30.1 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 an ADAT or SPDIF signal is applied to the Fireface 800, the corresponding input LED starts flashing. The unit indicates LOCK, i. e. a valid input signal (in case the signal is also in sync, the LED is constantly lit, see below).

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 [1]: The Fireface is set to 44.1 kHz internally (clock mode Master), and a mixing desk with ADAT output is connected to input ADAT1. The corresponding LED will show LOCK immediately, but usually the mixing desk's sample rate is generated internally (also Master), and thus slightly higher or lower than the Fireface'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 inputs, a simple LOCK is not sufficient. The above described problem can be solved elegantly by setting the Fireface from Master to AutoSync (its internal clock will then be the clock delivered by the mixing desk). But in case another, un-synchronous device is connected, there will again be a slight difference in the sample rate, and therefore clicks and drop outs.

In order to display those problems optically at the device, the Fireface includes **SyncCheck**[®]. It checks all clocks used for *synchronicity*. If they are not synchronous to each other (i. e. absolutely identical), the SYNC LED of the asynchronous input flashes. In case they are completely synchronous, all LEDs are constantly lit. In example 1 it would have been obvious that the LED ADAT 1 kept on flashing after connecting the mixing desk.

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.

The same information is presented in the Fireface's Settings dialog. In the status display *SyncCheck* the state of all clocks is decoded and shown as simple text (No Lock, Lock, Sync).

30.2 Latency and Monitoring

The term **Zero Latency Monitoring** was 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 Fireface 800 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 μ s (0.000068 s), at 192 kHz only 15 μ s. The delay is valid for ADAT and SPDIF in the same way.

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 (loopback) via DA and AD then causes an offset of the newly recorded track of about 2 ms.

The exact delays of the Fireface 800 are:

Sample frequency kHz	44.1	48	88.2	96	176.4	192
AD (43.2 x 1/fs) ms	0.98	0.9	0.49	0.45		
AD (38.2 x 1/fs) ms					0.22	0.2
DA (43.5 x 1/fs) ms	0.99	0.9	0.49	0.45	0.25	0.23

Buffer Size (Latency)

This option in the Settings dialog defines the size of the buffers for the audio data used in ASIO (see also chapter 8.1 and 9.4). A setting of 64 samples at 44.1 kHz causes a latency of 1.5 ms, for record and playback each. In the above example this latency stayed completely unnoticed, the newly recorded track was shifted only by the amount of the AD- and DA-converter's latency. The reason for this 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. Since the software can not determine the amount of external latency, the AD-/DA-converters etc. are not part of the automatic correction. The user has to add an offset in the software manually to compensate such delays.

Safety Buffer

FireWire audio differs significantly from RME's previous DMA technology. DMA access is not possible at all with FireWire. To be able to transmit audio reliably at lower latencies, FireWire requires a new concept – the *Safety Buffer*. The Fireface 800 uses a fixed additional buffer of 64 samples (on the playback side only), which is added to the current buffer size. The main advantage is the ability to use lowest latency at highest CPU loads. Furthermore, the fixed buffer does not add to the latency jitter (see Tech Info), the subjective timing is extraordinary. Disadvantage: the safety buffer's 64 samples needs to be considered when compensating offsets manually.

30.3 DS - Double Speed

When activating the *Double Speed* mode the Fireface 800 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* in connection with the ADAT format. The DTRS recorder DA-98HR by Tascam also uses this technique, which is called *Dual Line* here.

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.

Because the ADAT interface does not allow for sampling frequencies above 48 kHz (a limitation of the interface hardware), the Fireface 800 automatically uses the described *Sample Split* method in DS mode. One channel's data is distributed to two channels according to the following table:

Analog In	1	2	3	4	5	6	7	8
DS Signal	1/2	3/4	5/6	7/8	1/2	3/4	5/6	7/8
Port	ADAT1	ADAT1	ADAT1	ADAT1	ADAT2	ADAT2	ADAT2	ADAT2

As the transmission of double rate signals is done at standard sample rate (Single Speed), the ADAT outputs still deliver 44.1 kHz or 48 kHz.

30.4 QS – Quad Speed

Due to the small number of available devices that use sample rates up to 192 kHz, but even more due to a missing real world application (CD...), Quad Speed has had no broad success so far. An implementation of the ADAT format as double S/MUX would result in only two channels per optical output. Devices using this method are not known to us, so we decided to do without this format.

The AES outputs provide 192 kHz as Single Wire only.

30.5 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 V _{ss}	0.2 V up to 0.5 V _{ss}
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.

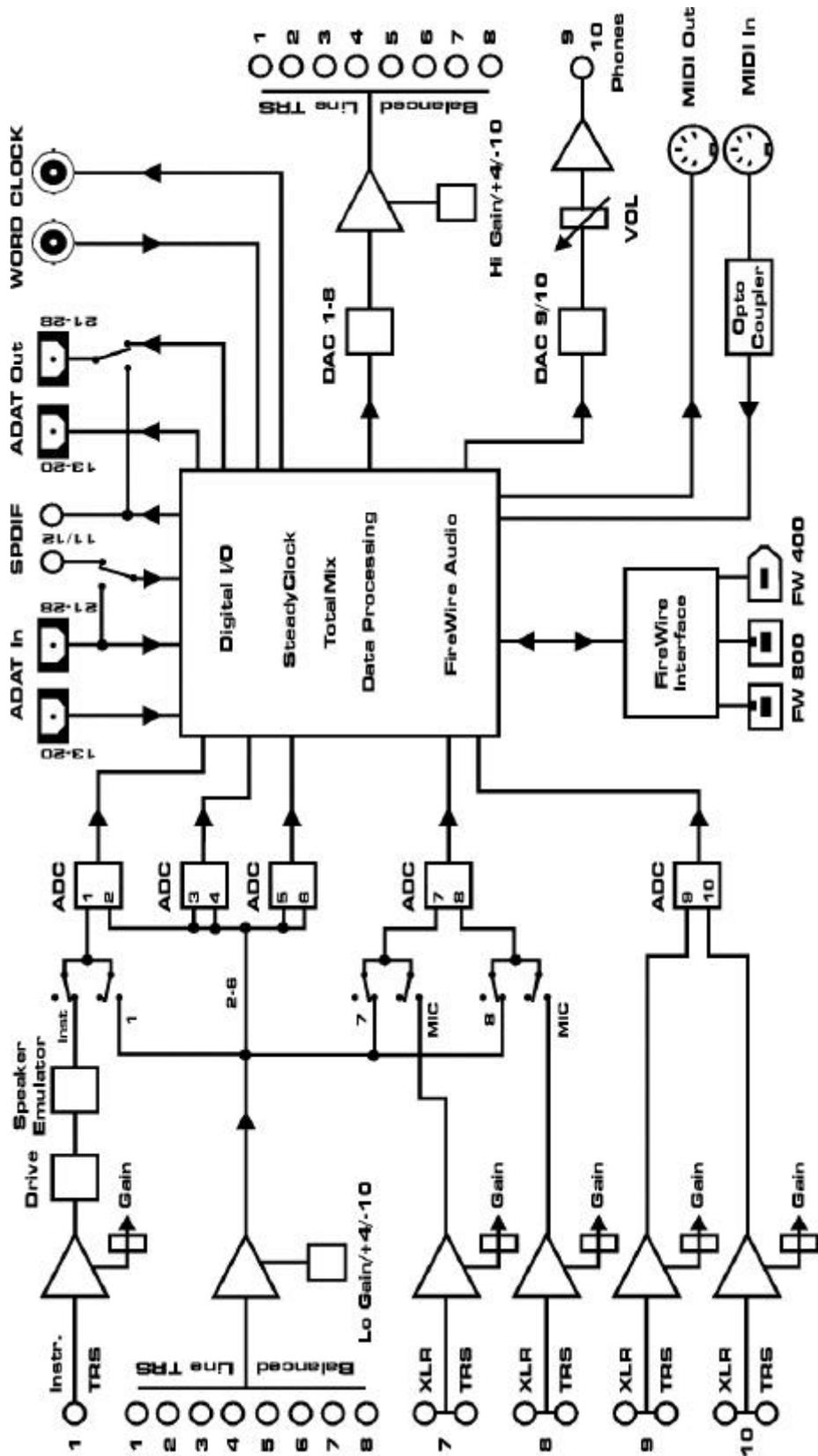
Byte	Mode	Bit 0	1	2	3	4	5	6	7
0	Pro	P/C	Audio?	Emphasis			Locked	Sample Freq.	
0	Con	P/C	Audio?	Copy	Emphasis			Mode	

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. This has actually happened in the past, but today it would be more likely to have been implemented on purpose, to force the customer to buy a more expensive device.

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

31. Diagrams

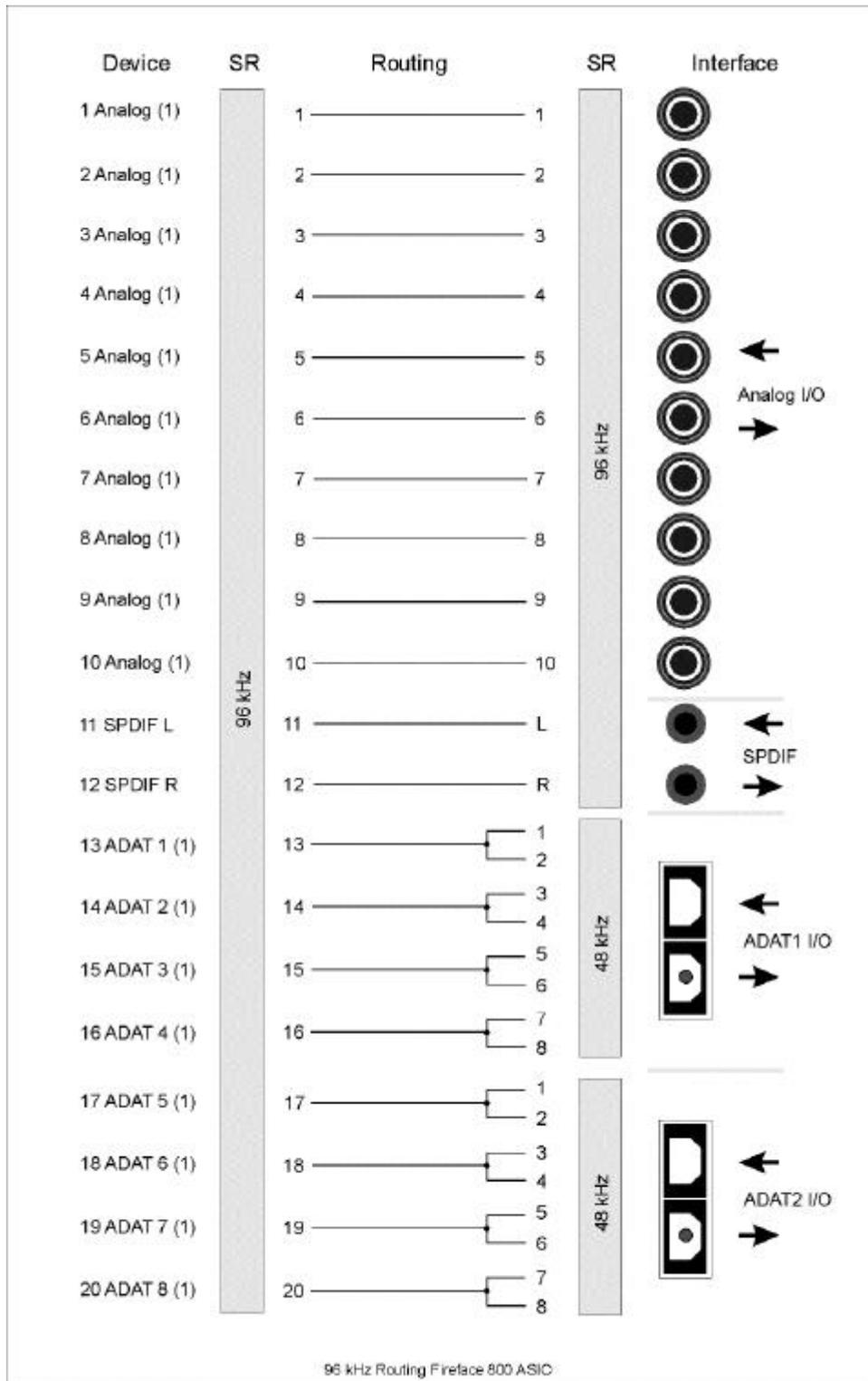
31.1 Block Diagram Fireface 800



31.2 Channel Routing ASIO at 96 kHz

This diagram shows the signal paths in ASIO double speed mode (88.2 / 96 kHz). The devices available via the ASIO driver have been designed to avoid conflicts in normal operation, which is why the second ADAT device has been omitted. Signal routing is identical for record and playback.

Device: Device name in the audio application SR: Sample Rate
 Device name code: Channel in ASIO host, interface, (unit number)

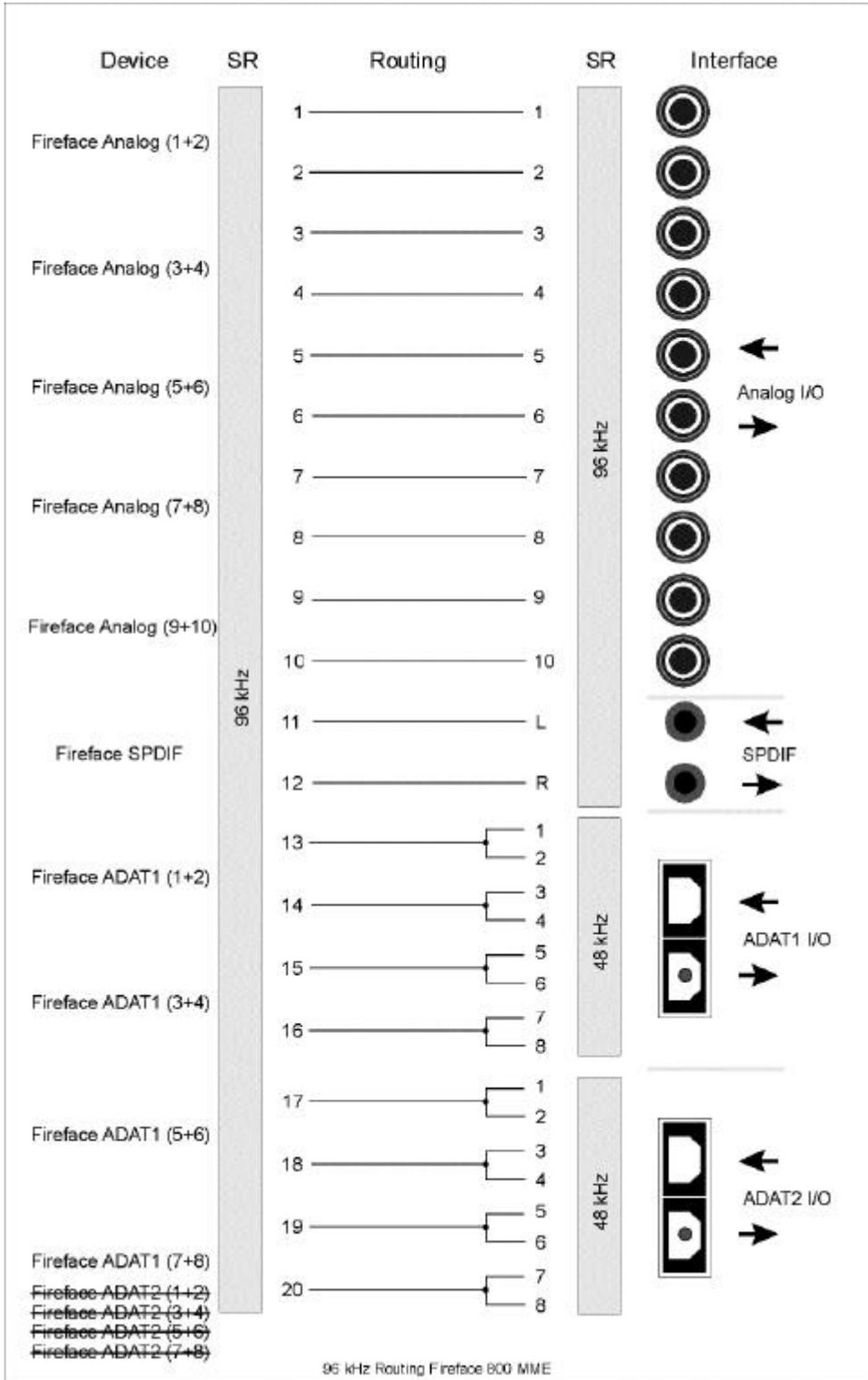


31.3 Channel Routing MME at 96 kHz

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

Device: Device name in the audio application

SR: Sample Rate



31.4 Connector Pinouts

TRS jacks of analog input / output

The stereo ¼" TRS jacks of the analog inputs and outputs are wired according to international standards:

Tip = + (hot)
Ring = - (cold)
Sleeve = GND

The servo balanced input and output circuitry allows to use monaural TS jacks (unbalanced) with no loss in level. This is the same as when using a TRS-jack with ring connected to ground.

XLR jacks of analog inputs

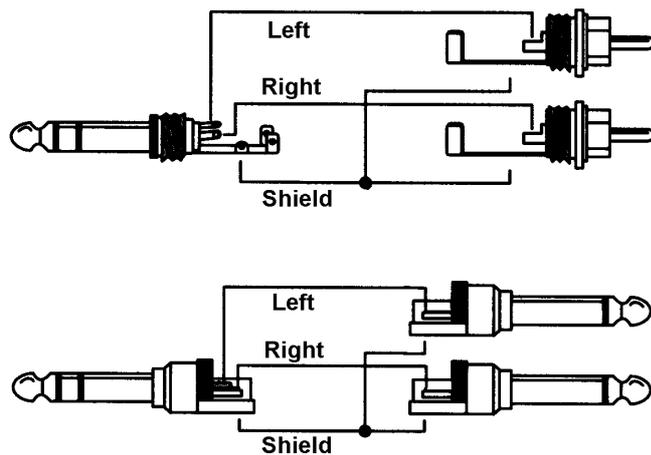
The XLR jacks are wired according to international standards:

1 = GND (shield)
2 = + (hot)
3 = - (cold)

TRS Phones jack

The analog monitor output on the front is accessible through a stereo ¼" TRS jack. This allows a direct connection of headphones. In case the output should operate as Line output, an adapter TRS plug to RCA phono plugs, or TRS plug to TS plugs is required.

The pin assignment follows international standards. The left channel is connected to the tip, the right channel to the ring of the TRS jack/plug.



32. 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.