
▶	Important Safety Instructions	4
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▶ **General**

1	Introduction	6
2	Package Contents	6
3	Brief Description and Characteristics	6
4	First Usage – Quick Start	
4.1	Controls - Connectors - Displays	7
4.2	Quick Start	9
5	Accessories	10
6	Warranty	11
7	Appendix	11

▶ **Usage and Operation**

8	Front Panel Controls	
8.1	Input	14
8.2	Limiter.....	14
8.3	Input State	15
8.4	Meter	15
8.5	Remote.....	15
8.6	Clock Section	15
8.7	Digital Input	16
8.8	Output State	16
8.9	Set (Rotary Encoder)	16
8.10	Output.....	17
9	The Setup Menu	
9.1	Exit	17
9.2	Global Output Level	17
9.3	Input Trim	18
9.4	Output Trim	18
9.5	ID (Id)	18
9.6	Bank (bA)	19
9.7	Auto ID (Au)	19
9.8	Delay Compensation (dC).....	19
9.9	Follow Clock (FC).....	20
9.10	Word Clock Out (Co).....	20
9.11	Trim Enable (TE).....	21
9.12	Global Output Level Enable (GE)	21
9.13	Direct Level (dL).....	21
9.14	Effects (FX)	21
9.15	Digital to Digital (dd).....	22
9.16	Digital Patch Mode	23
9.17	Dark and Lock Keys.....	24
10	Remote Control	
10.1	Hardware Remote	25
10.2	MIDI.....	25
10.3	MIDI over MADI.....	25
10.4	Remote Control Software.....	26

▶ **Inputs and Outputs**

11	Analog Inputs / Outputs	
11.1	Line In	30
11.2	Line Out	31
12	Digital Inputs / Outputs	
12.1	AES / EBU	32
12.2	ADAT Optical	33
12.3	I64 MADI Card	34
12.4	Differences I64 MADI Card / ADI-642	35
13	Word Clock	
13.1	Word Clock Input and Output	36
13.2	Technical Description and Background	37
13.3	Cables and Termination.....	38
14	MIDI	39
15	Remote	39

▶ **Technical Reference**

16	Technical Specifications	
16.1	Analog	42
16.2	Digital Inputs	43
16.3	Digital Outputs	44
16.4	Digital	44
16.5	MIDI	44
16.6	General	45
16.7	Firmware	45
16.8	MADI User Bit Chart	45
16.9	Connector Pinouts	46
17	Technical Background	
17.1	Terminology	48
17.2	Lock and SyncCheck	49
17.3	Latency and Monitoring	50
17.4	DS – Double Speed	51
17.5	QS – Quad Speed	51
17.6	AES/EBU – SPDIF	52
17.7	Signal to Noise ratio in DS / QS Operation.....	53
17.8	MADI Basics	54
17.9	SteadyClock.....	55
18	Block Diagram	56
19	MIDI Implementation ADI-8 QS	
19.1	Basic SysEx Format	57
19.2	Message Types	57
19.3	Table	58

Important Safety Instructions



ATTENTION! Do not open chassis – risk of electric shock

The unit has non-isolated live parts inside. No user serviceable parts inside. Refer service to qualified service personnel.



Mains

- The device must be earthed – never use it without proper grounding
- Do not use defective power cords
- Operation of the device is limited to the manual
- Use same type of fuse only



To reduce the risk of fire or electric shock do not expose this device to rain or moisture. Prevent moisture and water from entering the device. Never leave a pot with liquid on top of the device. Do not use this product near water, i. e. swimming pool, bathtub or wet basement. Danger of condensation inside – don't turn on before the device has reached room temperature.



Installation

Surface may become hot during operation – ensure sufficient ventilation. Avoid direct sun light and do not place it near other sources of heat, like radiators or stoves. When mounting in a rack, leave some space between this device and others for ventilation.



Unauthorized servicing/repair voids warranty. Only use accessories specified by the manufacturer.



Read the manual completely. It includes all information necessary to use and operate this device.

User's Guide



ADI-8 QS

▶ General

1. Introduction

RME's ADI-8 QS is an 8-channel high-end AD/DA converter with a truly unique feature set. The device combines excellent analog circuit design with the latest generation of outstanding low latency AD/DA converter chips. Along with its integrated SteadyClock, the ADI-8 QS offers AD and DA conversion of the highest quality.

When developing the ADI-8 QS we used all our experience, and also the experience of our customers, to create a unique, excellent and high-quality unit. Although the ADI-8 QS is built to highest reference standards, it still offers the typical, outstanding price/performance ratio known from RME. ADI-8 QS features will excite you – but even more the performance and smartness that it shows in solving any given task. Have fun!

2. Package Contents

Please check that your ADI-8 QS package contains each of the following:

- ADI-8 QS
- Power cord
- Manual
- RME Driver CD
- 1 optical cable (TOSLINK), 2 m
- Desktop remote control

3. Brief Description and Characteristics

The ADI-8 QS is an 8-channel hi-end AD and DA converter in reference quality, fully remote controllable. In a standard 19" box with 1 unit height the device offers numerous extraordinary features like Intelligent Clock Control (ICC), SyncCheck[®], SteadyClock, analog and digital Limiter, four hardware reference levels up to +24 dBu, AES/EBU and ADAT I/O, optional MADI I/O, 192 kHz sample rate, MIDI remote control, digital input and output trimming for full level calibration, calibration, volume control for all 8 analog outputs, either separately or globally, and a digital thru-mode.

- 8-channel AD converter, full symmetrical design, 117 dBA
- 8-channel DA converter, DC coupled design, double balanced output, 120 dBA
- Low latency conversion: less than 12 samples delay
- 4 x AES/EBU I/O per D-sub, 8 channels @ 192 kHz
- 2 x ADAT I/O, 8 channels @ 96 kHz
- Optional MADI I/O (164 MADI Card)
- Word clock input and output
- MIDI I/O
- Digital input/output trim over a range of 6 dB per I/O
- AD-conversion includes switchable analog and digital Limiter
- 2 x 8-channel level meter with 7 LEDs per channel
- Noise suppression on power-on and power-off at the analog outputs
- Included Hardware remote for store/recall, volume and DIM
- Fully remote controllable via MIDI and MADI

4. First Usage – Quick Start

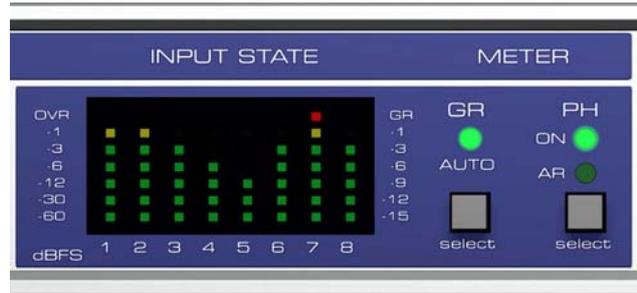
4.1 Controls - Connectors - Displays

The front of the ADI-8 QS features 16 LED level meters, nine select keys, one rotary encoder, and 30 LEDs providing a detailed status display.

The analog input sensitivity can be changed in the area **INPUT**. Choosing **DIG** with the select key, the digital input signal is routed to the digital outputs, with the AD-conversion being deactivated.

The 8-channel **LIMITER** is available as analog (**MODE ANA**), digital (**MODE DIG**) or combined (both LEDs lit) function.

The 8 level meter of the **INPUT STATE** show the digital value (dBFS) of the input level per channel. With activated **METER GR AUTO** the amount of Gain Reduction is shown in dB as soon as the Limiter becomes active. **METER PH** provides a Peak Hold function, either permanently (**ON**) or switched off by Auto Reset (**AR**) after 3 seconds.



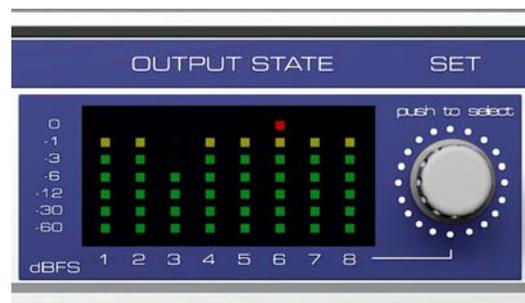
When the Setup menu is active, the according parameter values are displayed in this level meter window.

REMOTE defines the source for MIDI remote control operation (Option Slot/MADI or DIN jack).

In the **CLOCK SECTION** the current clock reference and frequency multiplier is chosen.

The digital input signal of the DA-converters is chosen by pushing the **DIGITAL INPUT** button.

The 8 level meter of the **OUTPUT STATE** show the digital input level per channel as digital value (dBFS). **METER PH** provides a Peak Hold function, either permanently (**ON**) or switched off by Auto Reset (**AR**) after 3 seconds.



When the Setup menu is active, all available options are displayed in this level meter window.

Pushing the knob **SET** activates the Setup Menu. The rotary encoder is a highly intuitive multifunctional device. It is used to set the trim values, the volume, and various options in the Setup menu.

The key **ANALOG OUT** defines the analog reference level at the analog outputs. This level is reached at full scale of the AD-converters, thus matching the front panel level meter's level indication.

The rear panel of the ADI-8 QS has eight analog inputs, eight analog outputs, mains power, a connector for the included Hardware Remote, MIDI I/O, word clock I/O, the I64 Option Slot, and all digital inputs and outputs (AES/ADAT).

ANALOG INPUTS: Eight balanced **Line inputs**, 1/4" TRS inputs and D-sub.

ANALOG OUTPUTS: Eight balanced **Line outputs**, 1/4" TRS outputs and D-sub.

AES I/O (25-pin D-sub): The D-sub connector provides four AES/EBU outputs (AD signals) and four AES/EBU inputs (DA signals). The 25 pin D-sub connector is wired according to the widely spread Tascam standard (pinout see chapter 16.9). The AES I/Os are transformer-coupled. The high sensitivity type input accepts all common digital sources, even SPDIF.

ADAT I/O MAIN (TOSLINK): Standard ADAT optical port.

ADAT I/O AUX (TOSLINK): Transmits further channels at activated sample multiplexing.



WORD IN (BNC): A push switch allows to activate internal termination (75 Ohms).

WORD OUT (BNC): Standard word clock output.

MIDI I/O (5-pin DIN): MIDI input and output via 5-pin DIN jacks. Used to remote control the ADI-8 QS, and – with installed MADI Card - to transmit MIDI data via MADI.

REMOTE (Mini DIN): Jack to connect the included Hardware Remote.

IEC receptacle for power connection. The specially developed, internal hi-performance switch mode power supply lets the ADI-8 QS operate in the range of 100V to 240V AC. It is short-circuit-proof, has an integrated line-filter, is fully regulated against voltage fluctuations, and suppresses mains interference.

With installed **I64 MADI Card**:

MADI I/O optical: Standard MADI ports.

MADI I/O coaxial (BNC): Standard MADI ports.



4.2 Quick Start

After connection of all cables and power-on of the device, the configuration of the ADI-8 QS begins in the CLOCK section. Choose a clock source and a sample rate.

Connect the TRS-jacks or the D-sub connector with the analog signal source. Change the input sensitivity by pressing INPUT until the input level is sufficient to avoid noisy operation. Try to achieve an optimum input level by adjusting the source itself. Raise the source's output level until the red Over LED flashes at the loudest parts of the signal, then reduce the level a bit until no more overs are detected.

The analog line inputs of the ADI-8 QS can be accessed by using D-sub (for an optional XLR multicore) and 1/4" TRS jacks. Both are internally connected, so not operational at the same time. The electronic input stage is built in a servo balanced design which handles unbalanced (mono TS) and balanced (XLR, stereo TRS jacks) correctly, automatically adjusting the level reference.

When switched on for the first time the ADI-8 DS starts in a default mode which should be suitable for most applications:

- AD/DA conversion in master mode (CLOCK INTERNAL)
- Sample rate 48 kHz
- Input ADAT optical
- Level reference +19 dBu

On the DA-side just choose the desired digital input by pressing DIGITAL INPUT. A coarse correction of the analog output level can be done by pressing the OUTPUT button.

The ADI-8 QS stores all current settings and automatically activates them when the device is turned on.

5. Accessories

RME offers several optional components for the ADI-8 QS:

Part Number	Description
OK0050	Optical cable, Toslink, 0.5 m (1.7 ft)
OK0100	Optical cable, Toslink, 1 m (3.3 ft)
OK0200	Optical cable, Toslink, 2 m (6.6 ft)
OK0300	Optical cable, Toslink, 3 m (9.9 ft)
OK0500	Optical cable, Toslink, 5 m (16.4 ft)
OK1000	Optical cable, Toslink, 10 m (32.8 ft)
BO25MXLR4M4F1PRO	Digital Breakout Cable Pro, AES/EBU 25-pin D-sub to 4 x XLR male + 4 x XLR female, 1m (3.3 ft)
BO25MXLR4M4F3PRO	same, 3 m (9.9 ft)
BO25MXLR4M4F6PRO	same, 6 m (19.8 ft)
BO25M25M1PRO	Digital D-sub Cable Pro, AES/EBU, 25-pin D-sub to 25-pin D-sub, 1m (3.3 ft)
BO25M25M3PRO	same, 3m (9.9 ft)
BO25M25M6PRO	same, 6m (19.8 ft)
BO25MXLR8M3	Analog Breakout Cable, 25-pin D-sub to 8 x XLR male, 3 m (9.9 ft)
BO25MXLR8M6	same, 6 m (19.8 ft)
BO25MXLR8M10	same, 10 m (33 ft)
BO25MXLR8F3	Analog Breakout Cable 25-pin D-sub to 8 x XLR female, 3 m (9.9 ft)
BO25MXLR8F6	same, 6 m (19.8 ft)
BO25MXLR8F10	same, 10 m (33 ft)
I64 MADi Card	MADi module featuring thru-input, Delay Compensation, Auto ID, MIDI over MADi, remote via MADi
BOB32	BOB-32, Universal breakout box, 19" 1 Unit height. The professional digital AES/EBU breakout solution

6. Warranty

Each individual ADI-8 QS undergoes comprehensive quality control and a complete test at IMM before shipping. The usage of high grade components allows us to offer a full two year warranty. We accept a copy of the sales receipt as valid warranty legitimation.

If you suspect that your product is faulty, please contact your local retailer.

Audio AG grants a limited manufacturer warranty of 6 months from the day of invoice showing the date of sale. The length of the warranty period is different per country. Please contact your local distributor for extended warranty information and service. Note that each country may have regional specific warranty implications.

In any case 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.

No warranty service is provided when the product is not returned to the local distributor in the region where the product had been originally shipped.

Audio AG does not accept claims for damages of any kind, especially consequential damage. Liability is limited to the value of the ADI-8 QS. The general terms of business drawn up by Audio AG apply at all times.

7. Appendix

RME news and further information can be found on our website:

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

Distributor: Audio AG, Am Pfanderling 60, D-85778 Haimhausen, Tel.: (49) 08133 / 918170

Manufacturer:

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

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CE / FCC Compliance

CE

This device has been tested and found to comply with the limits of the European Council Directive on the approximation of the laws of the member states relating to electromagnetic compatibility according to RL89/336/EWG and RL73/23/EWG.

FCC

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

RoHS

This product has been soldered lead-free and fulfills the requirements of the RoHS directive.

ISO 9001

This product has been manufactured under ISO 9001 quality management. The manufacturer, IMM Elektronik GmbH, is also certified for ISO 14001 (Environment) and ISO 13485 (medical devices).

Note on Disposal

According to the guide line RL2002/96/EG (WEEE – Directive on Waste Electrical and Electronic Equipment), valid for all european countries, this product has to be recycled at the end of its lifetime.

In case a disposal of electronic waste is not possible, the recycling can also be done by IMM Elektronik GmbH, the manufacturer of the ADI-8 QS.

For this the device has to be sent **free to the door** to:

IMM Elektronik GmbH
Leipziger Straße 32
D-09648 Mittweida
Germany



Shipments not prepaid will be rejected and returned on the original sender's costs.

The included hardware remote is a product of RME Trading Ltd., Hong Kong. In case a disposal of electronic waste is not possible, the recycling can be done by Audio AG. For this the device has to be sent **free to the door** to:

Audio AG
Am Pfanderling 60
D-85778 Haimhausen
Germany

Shipments not prepaid will be rejected and returned on the original sender's costs.

User's Guide



ADI-8 QS

▶ Usage and Operation

8. Front Panel Controls

8.1 Input

In the **INPUT** area, the select key causes a change of the analog input sensitivity, referenced to digital full scale level (0 dBFS) of the AD-converters. The choices are: **+4.2 dBu** (-10 dBV compatible), **+13 dBu**, **+19 dBu** and **+24 dBu**. As the adaptation is performed in the analog domain, the ADI-8 QS achieves the highest possible values for Signal to Noise ratio and distortion in any setting.



After selecting **DIG**, the digital input signal is routed to the digital outputs, with the AD-conversion being deactivated. The level meters now show the level of the digital input signal.

8.2 Limiter

The **LIMITER** is available as analog function (**MODE ANA**), digital function (**MODE DIG**), or combined (both LEDs lit).

Analog Limiter

For the ADI-8 QS, RME developed a new fully symmetrical circuit design with digitally controlled release time. While the analog limiter is constantly within the signal path, it operates fully transparent as long as there is no control signal generated, so does not cause noise or distortion. This special feature made it possible to define a threshold of +3 dB in deactivated mode, preventing an overload of more than 3 dB of the AD-converters. This avoids not only extreme overload effects, but also prevents the AD-converters from being damaged.



Both the fully symmetrical design and the digitally controlled release time contribute to the reduction of the unavoidable distortion of a peak limiter (linearity errors of the control elements, generation of the control voltage). The distortion value of about 0.3% stays the same at any level and any frequency.

Another exceptional feature is the ability to handle extremely high levels. At the level references +24 dBu, +19 dBu and +13 dBu up to +30 dBu can be fed, the signal is still limited correctly (-3 dBFS), THD is still 0.3%. At the setting +4.2 dBu the maximum unclipped input level is +24 dBu. So there is more than 20 dB overload possible!

Note: As with all such basic designs, increasing dynamic ratio and gain reduction causes audible pumping effects.

Digital Limiter

The digital limiter is a non linear compressor. The input signal is processed with a special amplitude function. The resulting third order harmonics are widely masked by the original signal.

The digital limiter allows for additional headroom of up to 5 dB without audible clipping (clipping: brickwall limiting leading to high distortion). It also allows for an increase of the loudness without increasing the maximum peak level, similar to the effect known as Maximizer. Unlike the analog limiter there is no attack and release time and therefore no pumping effects.

The strength of the limiter effect can be adjusted in the Setup menu in steps of 1.5, 3, 4 and 5 dB, indicating the maximum level or loudness gain (chapter 9.14).

8.3 Input State

The 8 level meter of the **INPUT STATE** show the digital value (dBFS) of the input level per channel. The level data incorporate Input Trim and Digital Limiter. The red LED OVR is lit 0.2 dB before full scale level (-0.2 dBFS).

With activated **METER GR AUTO** the amount of Gain Reduction is shown in dB as bar from top to bottom, as soon as the Limiter becomes active. In case of overload a display is even performed when the Limiter has not been activated, because the Limiter is indeed always active, operating at a threshold of +3 dB.

When the Setup menu is active, the according parameter values are displayed in this level meter window.

8.4 Meter

METER PH provides a Peak Hold function for all level meters, either permanently (**ON**) or switched off by Auto Reset (**AR**) after 3 seconds.

In mode ON, a reset (erasing the current peak values) is issued by a single push of the PH button. The function itself is not changed.



8.5 Remote

REMOTE defines the source of MIDI remote control commands. Choices are the MIDI DIN jack, AES1 and the MAD1 input of the I64 MADI Card (Option Slot).

Note: Via MIDI remote control, all front panel controls can be locked (Lock Keys). An exception is the REMOTE key. In the Off state (no LED lit) Lock Keys is deactivated. Therefore a locking of all the controls can be revoked directly at the unit at any time.



8.6 Clock Section

The source and frequency of the unit's clock is configured in the **CLOCK SECTION**. The button **CLOCK** lets you step through the options external clock (Word, ADAT, AES, Option = MADI) and internal clock, and sets the sample rate for both an external and the internal clock.

WCK, AES, OPTN (Slave Mode)

Defines the corresponding input as clock reference. A missing or invalid clock source signal is indicated by flashing of the corresponding LED.

INT (Master Mode)

Activates the internal clock.



With a setting of INT (internal clock) it is mandatory that the clock rate of the sources is synchronous to the ADI-8 QS. Therefore the external device has to be synchronized to the ADI-8 QS word clock output or AES/ADAT/MADI output.

The ADI-8 QS thus has to be master, all devices connected to it must be slave. In order to avoid clicks and drop outs due to faulty or missing synchronicity, a special process called *SyncCheck* compares the incoming data and the ADI-8 QS internal clock. The SYNC state is indicated by a flashing (error) or turned off (OK) LED.

44.1, 48

Activates the internal clock at 44.1 kHz or 48 kHz.

DS, QS

With the DS LED additionally lit, the sample rate will be 88.2 or 96 kHz, with QS lit it will be 176.4 or 192 kHz.

A selection of DS and QS is also possible when using external clock (Slave). If the ADI-8 QS should operate at 192 kHz, but receives a synchronous word clock of 48 kHz, the button STATE allows to activate DS or QS mode. This way, AD-conversion and digital outputs are configured to operate in the frequency ranges Single Speed, Double Speed or Quad Speed.



Single Speed

All outputs carry a signal in the range of 32 kHz up to 48 kHz.

DS (Double Speed)

The AES outputs 1-8 carry a signal in the range of 64 kHz up to 96 kHz. ADAT and MADI stay at no higher than 48 kHz, with the data transmitted in the S/MUX format.

QS (Quad Speed)

The AES outputs 1-8 carry a signal in the range of 176.4 kHz up to 192 kHz. ADAT and MADI stay at no higher than 48 kHz, with the data transmitted in the S/MUX4 format. Therefore ADAT is limited to 4 channels (2 per optical output) in this mode.

8.7 Digital Input

The key **DIGITAL INPUT** defines the signal source of the DA-converters. This is also true for the mode *Digital to Digital*, where the chosen digital input signal is sent directly to the digital outputs.

Each input has its own SYNC LED. If a valid input signal is applied, SyncCheck is active automatically. SyncCheck takes the chosen clock (internal, external, etc.) as reference and compares it with the input clocks. Inputs which are not synchronous will be signalled by flashing of the corresponding SYNC LED.



8.8 Output State

The 8 level meter of the **OUTPUT STATE** show the digital input level per channel as digital value (dBFS). The display incorporates the settings of Global Output Level and Output Trim. Therefore overloads of the DA-converters will always be displayed.

When the Setup menu is active, all available options are displayed in this level meter window.

8.9 Set (Rotary Encoder)

The rotary encoder is a highly intuitive multifunctional device. It is used to set the trim values, the volume, and various options in the Setup menu.

Pushing the knob **SET** activates the Setup menu. Within the Setup menu a push on SET changes between left and right window.



8.10 Output

In the **OUTPUT** area, the select key causes a change of the analog output level, referenced to digital full scale level (0 dBFS) of the DA-converters. The choices are: **+4.2 dBu** (-10 dBV compatible), **+13 dBu**, **+19 dBu** and **+24 dBu**. As the adaptation is performed in the analog domain, the ADI-8 QS achieves the highest possible values for Signal to Noise ratio and distortion ratio in any setting.



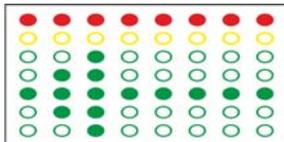
9. The Setup Menu

Some options and settings are seldom accessed and changed. These have been collected in the Setup menu. The Setup menu is entered by pushing the rotary encoder (SET) in both level meter windows. The right window shows all the options, the left window is used to configure them. In the currently active window all red LEDs are lit, similar to a cursor function.

The setup procedure is run as follows:

- Push SET one time. The level display vanishes, the cursor marks the right window.
- Turn SET left or right until the desired option shows up, for example *Digital to Digital*, shown as the letters d d.
- Push SET one time. The cursor jumps to the left window.
- Turn SET left or right until the desired parameter value shows up. In this example, d d can be activated (check symbol) or deactivated (no check symbol).
- Push SET one time to return to the options in the right window, or simply wait for 6 seconds. The last setting stays valid, the setup menu is left, and the level meters work again.

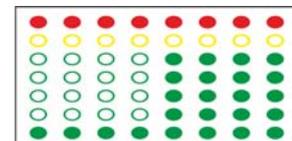
9.1 Exit



The Setup menu is exit automatically when no entries are received for about 6 seconds. A quicker is to use the option Exit. Selecting the arrow in the left window will exit the Setup menu immediately. All settings performed before are stored and stay valid.

9.2 Global Output Level

Ganged setting of all 8 analog output levels simultaneously. This function provides a global volume control, which is indispensable when doing surround monitoring. The included remote control box directly uses this function.



The level attenuation is calculated and performed in the digital domain. When using higher attenuation settings most of the time, the hardware output level (OUTPUT) should be changed to a lower setting to utilize the whole dynamic range of the DA-converters as much as possible.

There are 48 steps from 0 dB (no attenuation) down to -96 dB available. Each step is shown as one LED in the left window. The more LEDs light up, the higher is the output level. At 0 dB attenuation all 48 LEDs are lit. In the range of 0 dB down to -20 dB the attenuation takes place in steps of 1 dB.

When all LEDs of a column are turned off, the corresponding attenuation is is:

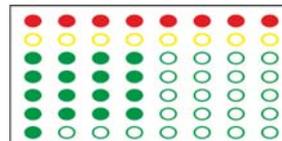
Channel 8	-6 dB	Channel 5	-28 dB	Channel 2	-78 dB
Channel 7	-12 dB	Channel 4	-40 dB	No LED	Mute
Channel 6	-18 dB	Channel 3	-54 dB		

When changing the Global Output Level the individual settings of Output Trim are maintained. So the gain of all channels can be increased or decreased without them losing their level relations.

9.3 Input Trim

Default: 0 dB

Available settings: 0 up to +6 dB in steps of 0.5 dB



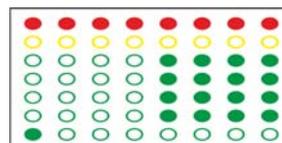
Additionally to the four hardware-based reference levels a digital amplification of up to +6 dB can be applied to any input channel. This allows for an individual and separate level calibration of the AD-conversion. At the same time the ADI-8 QS is provided with a near stepless input sensitivity from +7 dBu up to +24 dBu, adjustable in steps of 0.5 dB.

Naturally the digital amplification reduces the signal to noise ratio according to the current gain. At +6 dB the maximum SNR of 114 dB is reduced to 108 dB.

9.4 Output Trim

Default: 0 dB

Available settings: -6 dB up to +6 dB, in steps of 0.5 dB



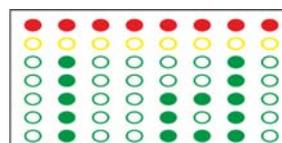
Additionally to the four hardware-based reference levels a digital gain change of +/- 6 dB can be applied to any output channel. This allows for an individual and separate level calibration of the DA-converter outputs, as required in a surround monitoring setup.

With a full scale signal (0 dBFS) positive values cause an overload of the DA-converter, but this is clearly shown on the output level meter. Using the ADI-8 QS as converter for surround monitoring, overloads are usually no issue. The typical operation range of the volume control is about -6 and -12 dB. With this there is enough headroom available for an amplification of up to 6 dB.

9.5 ID (Id)

Default: 1

Available settings: 1, 2, 3, 4, 5, 6, 7, 8



To remote control more than one ADI-8 QS each device can have its own ID, providing a separated remote control of multiple devices via a single MIDI channel.

164 MADI Card: The ID defines the 8-channel group within the MADI signal that is used to insert the device's audio data:

ID 1: channels 1-8	ID 4: channels 25-32	ID 7: channels 49-56
ID 2: channels 9-16	ID 5: channels 33-40	ID 8: channels 57-64
ID 3: channels 17-24	ID 6: channels 41-48	

When several ADI-8 QS, Micstasy or ADI-642 are connected via MADi, Auto ID normally takes care of the channel assignment (see chapter 9.7, Auto ID). In special cases, it may be desirable to set the ID manually, e.g. if the first device in a MADi chain does not support the Auto ID mode, or if a group of eight channels needs to be routed or processed in a particular way.

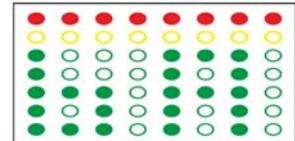
With activated mode *Digital to Digital* ID also defines which MADi channels are sent to the ADAT/AES outputs, see chapter 9.15.

When the ADI-8 QS is in mode *Auto ID Slave*, the automatically given ID is displayed by a single red LED lit, replacing the cursor display (all red LEDs lit). A manual setting of the ID is still possible, but is not active during slave mode.

9.6 Bank (bA)

Default: 1

Available settings: 1, 2, 3, 4, 5, 6, 7, 8

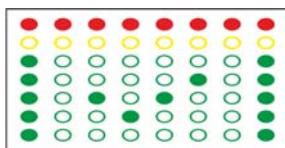
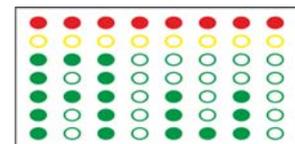


To remote control more than one ADI-8 QS each device can have its own ID, providing a separated remote control of multiple devices via a single MIDI channel (chapter 9.5). Additionally But there are not only 8 IDs, but 8 banks of 8 IDs each available.

9.7 Auto ID (Au)

Default: Off

Available settings: On, Off

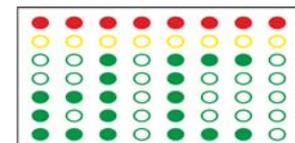


This option relates to the use of the I64 MADi Card. Multiple ADI-8 QS (and Micstasy, ADI-642, OctaMic XTC, see chapter 12.3) connected serially can assign consecutive IDs to themselves (see chapter 9.5). At the first device in the chain Auto ID is set to On (see picture to the left), all other devices automatically turn into slaves.

9.8 Delay Compensation (dC)

Default: Off

Available settings: On, Off



This option relates to the use of the I64 MADi Card. When multiple devices are connected serially, the MADi I/O of each ADI-8 QS causes a delay of 3 samples. Therefore at the MADi output of the last device, the data of all previous devices are delayed. At Double Speed the delay rises to 6 samples per unit, at Quad Speed to 12 samples.

The problem of this offset is solved by the function *Delay Compensation*. It delays the signals in a way that they are sample-synchronous in multi-device operation.



Attention: Delay Compensation has to be activated manually in each unit!

The following table lists the delay in samples from two up to eight units connected serially. When using four units, the data of the first unit are delayed by 9 samples to the last unit, the units 2 and 3 are delayed by 6 and 3 samples respectively. At Double Speed and Quad Speed the values rise. Please note that in Double Speed no more than four, in Quad Speed no more than two ADI-8 QS can be used serially with MADi.

Units	Delay	Delay DS	Delay QS	DC	DC DS	DC QS
2	3	6	12	21	18	12
3	6	12	-	21	18	-
4	9	18	-	21	18	-
5	12	-	-	21	-	-
6	15	-	-	21	-	-
7	18	-	-	21	-	-
8	21	-	-	21	-	-

21 samples @ 48 kHz
equal 437 µs.

18 samples @ 96 kHz
equal 187 µs.

12 samples @ 192
kHz equal 62.5 µs.

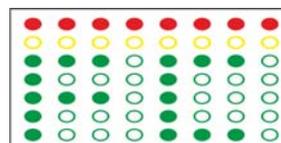
As shown in the table, activating DC causes a constant delay of 21 samples in Single Speed, no matter how many devices are connected serially. In Double Speed the delay is 18, in Quad Speed 12 samples. The in most cases slightly increased delay is outweighed by the big advantage of sample-aligned I/Os when using multiple units.

Delay Compensation always uses the worst case, the operation of 8 units, but delays the signals individually. The amount of the respective delay is solely derived from the current ID, no matter if it has been assigned manually or by Auto ID.

9.9 Follow Clock (FC)

Default: Off

Available settings: On, Off

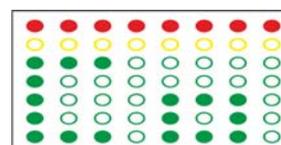


The signal at the word clock or AES input can be Single, Double or Quad Speed, the ADI-8 QS can use it in any case, independent of its current clock range. When activating the option *Follow Clock* the ADI-8 QS follows the input clock 1:1. At 96 kHz the DS LED will light up automatically, at 192 kHz the QS LED. This function is automatically deactivated for the I64 MADI Card and the ADAT input, as it is not possible to detect the sample rate range automatically.

9.10 Word Clock Out (Co)

Default: FS

Available settings: Si, FS



Si stands for Always Single Speed, FS for the current sample rate or Follow Sample rate. In default state (FS) the word clock output follows the current sample rate up to 192 kHz. Selecting Si causes the word clock output 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.

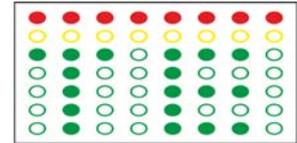
Note: A reliable sample-accurate synchronicity between multiple devices having digital interfaces in S/MUX technology (ADAT and MADI*) at sample rates in the Double Speed and Quad Speed range is only possible when all units are synced by Single Speed word clock. Because of the S/MUX method the device else can not know which of the incoming 2 (DS) or 4 (QS) word clock edges is the right one.

* With the ADI-8 QS this limitation is also valid for the AES output, as the device internally handles all the data streams in S/MUX mode.

9.11 Trim Enable (TE)

Default: On

Available settings: On, Off

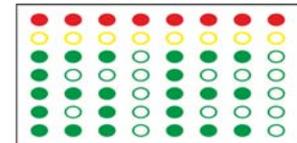


The settings of the level calibration per channel, Input Trim and Output Trim (see chapter 9.3 and 9.4), can be deactivated globally. The gain is then set to 0 dB. The individual settings are not lost, changing between the calibrated (On) and the linear setting (Off) is possible back and forth anytime.

9.12 Global Output Level Enable (GE)

Default: On

Available settings: On, Off



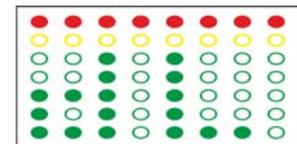
With the volume control deactivated the gain is fixed to 0 dB. Even if Direct Level is activated SET can not be used to change the volume.

Note: This setting is also valid for the included remote control box.

9.13 Direct Level (dL)

Default: Off

Available settings: On, Off



When Direct Level is activated, turning the rotary encoder immediately changes the global output level, in the same way as the included Hardware Remote. This option provides a comfortable setting of the volume at the unit, without the need to enter the according Setup menu first.

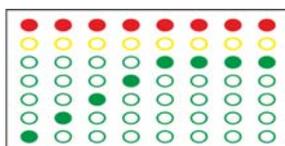
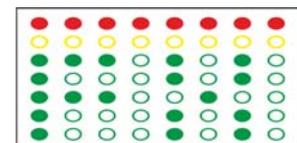
At each change the level meters are shortly switched off, displaying the current gain value instead.

Note: The included Hardware Remote is operational even when Direct Level is deactivated.

9.14 Effects (FX)

Default: 5

Available settings: 1.5, 3, 4, 5 dB

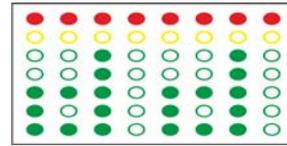


The digital limiter can be adjusted in four steps. The left LED display shows the I/O ratio curve of a limiter. The picture to the left shows the setting 1.5 dB, having the smallest effect. At 5 dB the curve is more horizontal, indicating a stronger influence on the input signal.

9.15 Digital to Digital (dd)

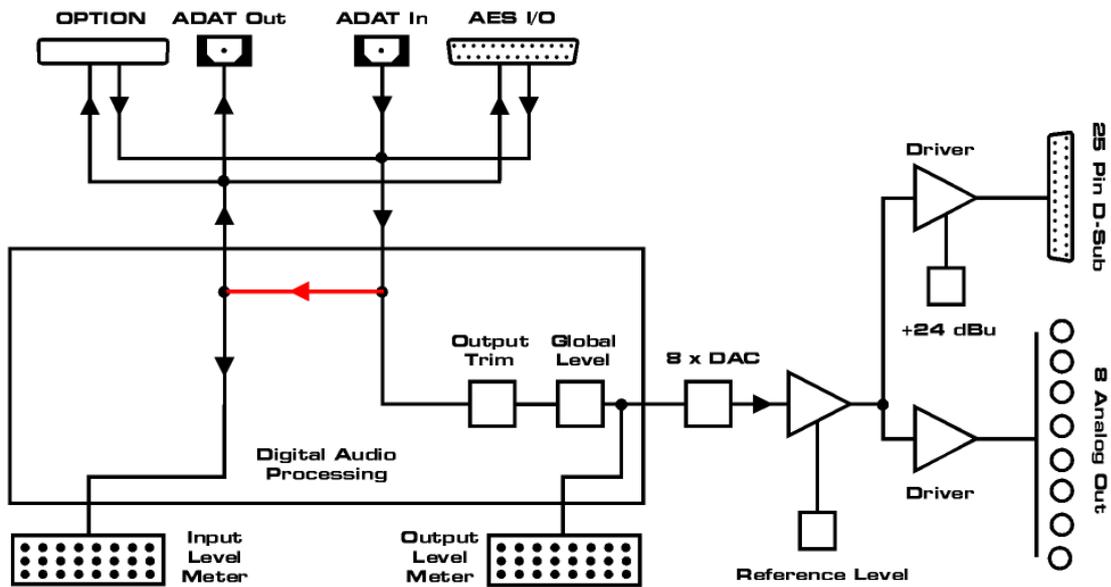
Default: Direct

Available settings: Direct, Effects

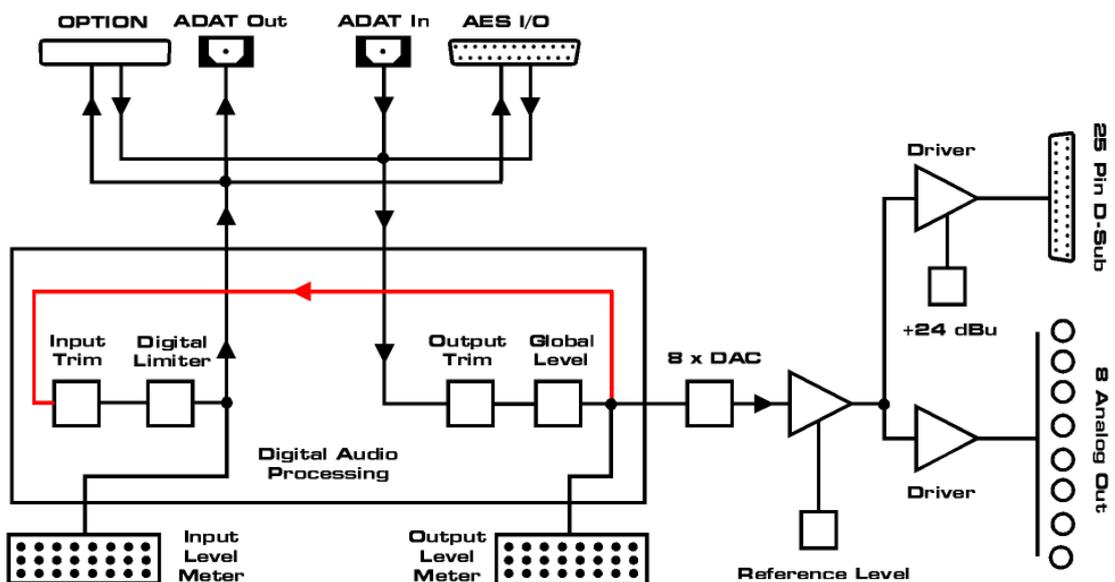


The function Digital to Digital can operate in two ways. In mode *Direct* the digital input signal is sent with bit-accuracy to the digital outputs. In mode *Effects* the digital input signal is first processed by the digital settings Output Trim and Global Level, then by the Input Trim settings, then – when activated – by the digital limiter, before finally reaching the digital outputs. The block diagrams show the signal flow for both modes.

Digital to Digital - Direct



Digital to Digital - Effects

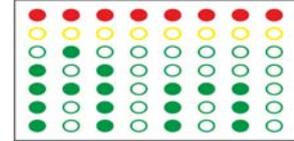


9.16 Digital Patch Mode

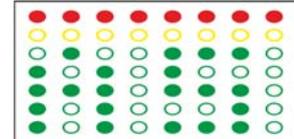
The symbol **o** stands for the Option Slot, MAD I, and will appear only when the ADI-8 QS has a firmware 3.0 or higher, and is equipped with an i64 MAD I card with two Sync LEDs (since late 2011). Then a *digital patch mode* is available, where up to 12 different outputs can be freely assigned to up to 12 inputs.

The number 12 is a result of dividing all inputs and outputs into 8-channel blocks:

An Analog, channels 1-8

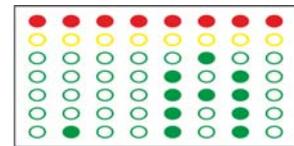
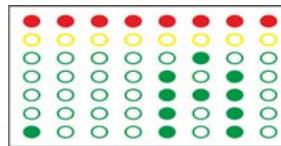


AS AES, channels 1-8

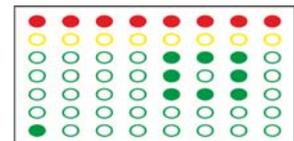


.A ADAT Main, channels 1-8

.A ADAT Aux, channels 1-8



- o (1)** MAD I channels 1-8
- o (2)** MAD I channels 9-16
- o (3)** MAD I channels 17-24
- o (4)** MAD I channels 25-32
- o (5)** MAD I channels 33-40
- o (6)** MAD I channels 41-48
- o (7)** MAD I channels 49-56
- o (8)** MAD I channels 57-64



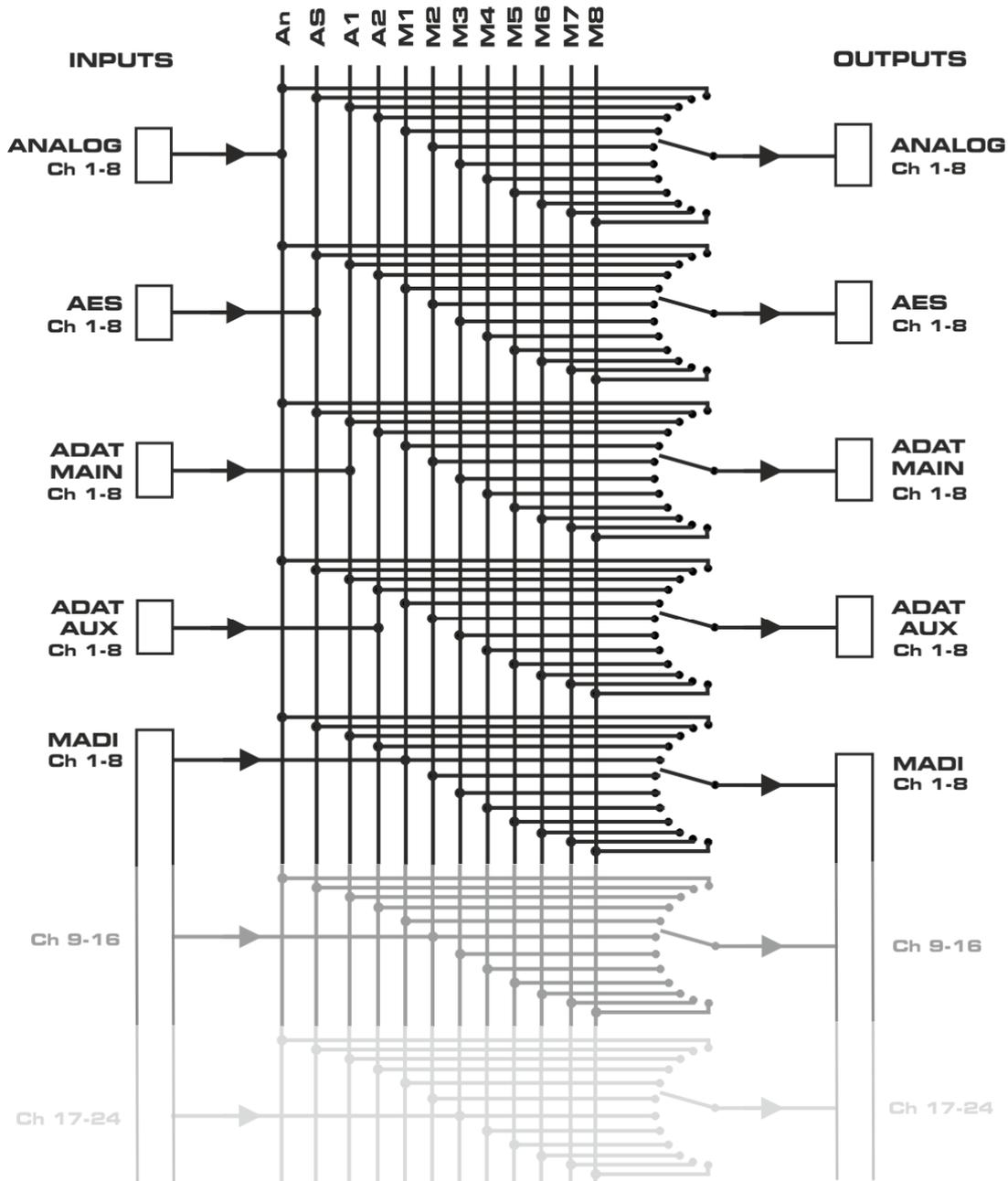
The bottom LED signals the current 8-channel output block. One input channel block can then be assigned to the current output block via the left LED display where all the above options are available as input choice. The MAD I output signal can therefore be freely rebuilt from the ADI-8 QS' up to 12 input blocks.

The digital patch mode is available for all hardware outputs. Further turning the encoder will make the analog outputs, ADAT 1 (Main) and ADAT 2 (Aux) as well as AES/EBU (same symbols as on the left) appear on the right side. Each of these output blocks can have one of the input blocks assigned to.

With this the QS can serve as converter from ADAT to AES and/or AES to ADAT. It can pass through ADAT to ADAT, with simultaneous analog monitoring, sort MAD I channel blocks freely or convert these to different formats, insert ADAT and/or AES into MAD I, serve as MAD I break-out box, and much more.

To **activate** or **deactivate** the **Digital Patch Mode** press and hold the key DIGITAL INPUT for 4 seconds. If active all LEDs of the DIGITAL INPUT section are lit constantly (valid input signal) or flash (none or not synchronous). The ADI-8 QS stores all settings of the routings.

The block diagram on the next page again shows the output-related selection of the available inputs, the division into blocks 1 to 7 (MAD I blocks 8 to 12 are not shown for better overview), and that the digital patch mode does not offer fully free channel-wise routing or mixing.

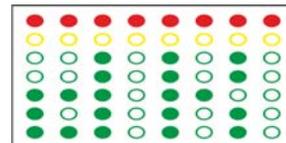


Digital Patch Mode ADI-8 QS

9.17 Dark and Lock Keys (dK)

Default: Off

Available settings: On, Off



Turns off all LEDs after a few seconds (stage mode), and disables all buttons and the encoder (lock keys mode). To temporarily reactivate all LEDs and controls press any button for 3 seconds.

10. Remote Control

10.1 Hardware Remote

The included Hardware Remote offers direct access to the most important functions needed in a typical DAW workplace:

- Turning the big knob changes the volume (Global Output Level)
- Pushing the big knob activates DIM (Global Output Level -20 dB)
- Pushing the PROG button for 2 seconds stores the current level setting
- After changing the volume, RECALL sets the stored level value again

Note: The Hardware Remote uses the function Global Output Level, see chapter 9.2. The volume can not be changed if this option is deactivated in the Setup menu.

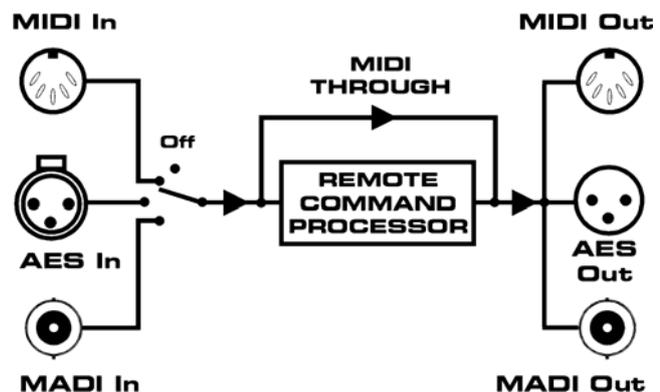
10.2 MIDI

The ADI-8 QS can be completely remote controlled via MIDI. It reacts on special SysEx commands. Furthermore, upon request it will report the complete device status, including all controls and LEDs on the front plate. Each ADI-8 QS can be programmed with its own ID, providing a separated remote control of multiple devices via a single MIDI channel. A description of the MIDI implementation is found in chapter 19.

The button REMOTE is used to select the source of the MIDI remote control commands: MIDI port, MADI port and Off. The latter is a safety function to prevent any setup-change by any MIDI signal.

With an installed I64 MADI Card the ADI-8 QS can also be remote controlled via MADI. Status information of the unit is sent to all outputs simultaneously. With an installed I64 MADI Card this includes embedded information at the MADI outputs (see 15.2, MIDI over MADI).

The diagram shows the signal flow of the MIDI data with all I/Os. The MIDI input signal reaches both the internal Remote Command Processor and the MIDI outputs. This *MIDI Through* function provides the base for a simplified serial MIDI cabling when using multiple ADI-8 QS. The same is true for remote control via MADI, where MIDI is passed through from device to device automatically via the serial MADI cabling.



10.3 MIDI over MADI

MADI allows for a transmission of 64 audio channels over long distances with a single line – perfect. So what about MIDI? Be it remote control commands or sequencer data, in practice only a single MADI line will not suffice. Therefore RME developed a *MIDI over MADI* technology. The data at the MIDI input are being included into the MADI signal invisibly, and can be collected at the MIDI output of another ADI-8 QS, ADI-6432, ADI-642, ADI-648, Micstasy or a HDSP MADI, at the other end of the MADI line.

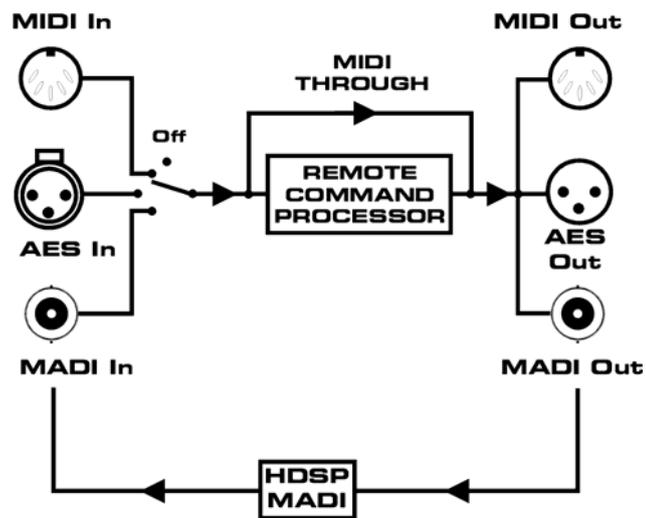
Technically every single MADI channel includes several additional bits, containing various information (Channel Status). RME uses the usually unused *User bit* of channel 56 (channel 28 in 96k frame mode) to transmit MIDI data invisibly within MADI, ensuring full compatibility.



Unlike other MADI-based devices from RME, the ADI-8 QS is no bi-directional MIDI to MADI converter, because only one direction is available for the transmission.

The block diagram shows the signal flow in a HDSP MADI-based remote control system. MIDI commands from a software on PC or Mac travel from the MADI Out of the HDSP MADI to the MADI In as well as to the MIDI Out and MADI Out of the ADI-8 QS.

Additional to the remote commands further MIDI data can be transmitted, available then at the DIN MIDI output jack of the ADI-8 QS. However, MIDI signals at the DIN input jack can not travel back to the computer. With input setting MIDI the other direction is active. MIDI data travel from the MADI out to the computer, but not back from the computer via MADI to the ADI-8 QS.



10.4 Remote Control Software

A software can be downloaded for free from the RME website. It can use any existing MIDI port within the system to perform remote control and status requests of any number of ADI-8 QS via a simple mouse click. Most appealing is the combination with a HDSP MADI (PCI card), offering a direct control of the ADI-8 QS via MADI. The remote control software then uses a virtual MIDI port of the card (port 3), which directly sends and receives MIDI data via MADI.

Software download: <http://www.rme-audio.de>, Downloads, MIDI Remote

Brief description of the Software ADI-8 QS MIDI Remote for Windows and Mac OS

The unique multi-window concept of the software MIDI Remote allows for a simultaneous usage and configuration not only of multiple ADI-8 QS, but also of all other supported devices, even in mixed setups. The function **ADI-8 QS** has been extended compared to the unit. All settings of the Setup menu are directly accessible, the gain reduction is shown on separate meters etc. Names can be assigned to all channels and all devices.

The program includes an extensive english online help (F1). After the start first choose the function **ADI-8 QS** in the menu **Functions** (also available via F4).

Then select a MIDI input and output via **Options - MIDI I/O Setup**.

Using the command **Options – Start/Stop MIDI I/O** the communication with the ADI-8 QS is started. The top line of the program window shows the current state, like selected ID, Online / No Response / Offline etc.

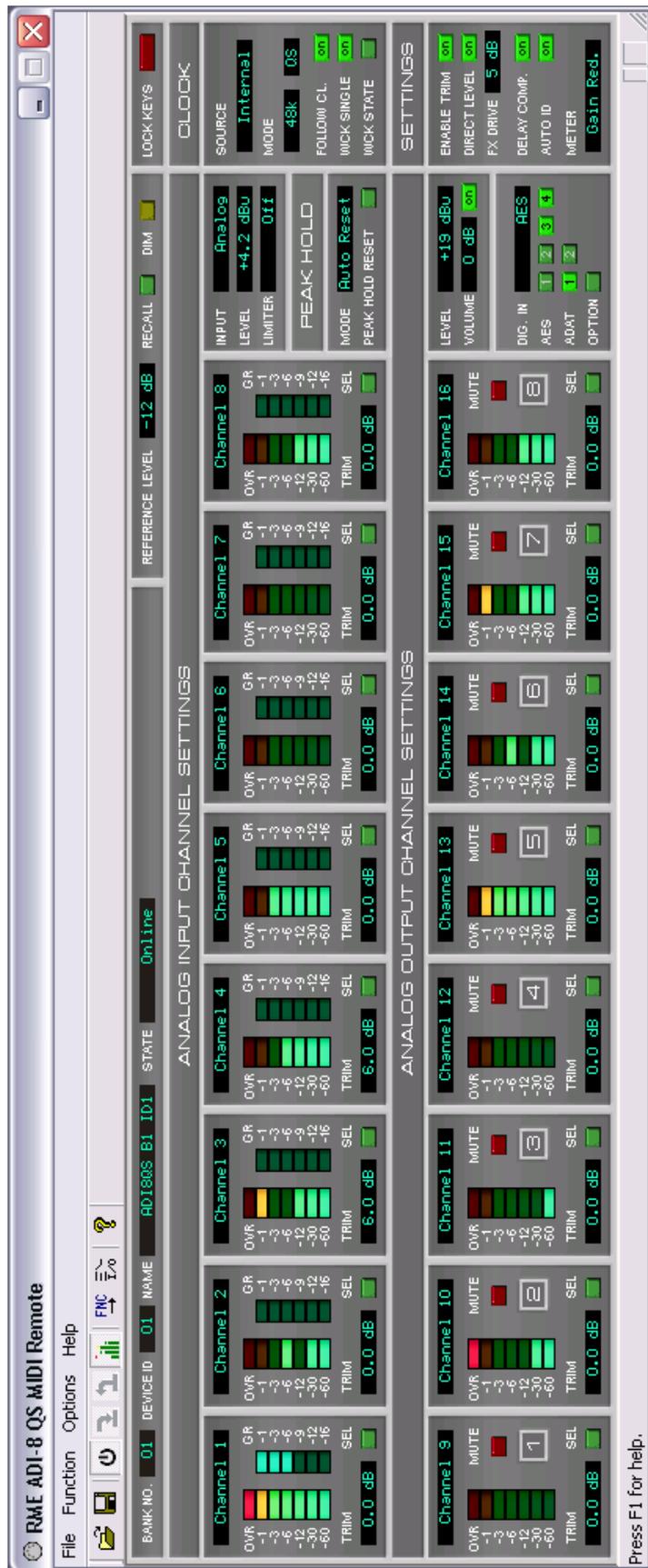
Save Workspace as allows to store complete setups including all windows in a file, and to re-load them at any time.

Send Single Set of Data allows an offline configuration of the ADI-8 QS, and uses a single one-time transmission of all settings.

Via MIDI remote control, all front panel controls of the ADI-8 QS can be locked (**Lock Keys**). An exception is the REMOTE key. In Off-state Lock Keys is deactivated. Therefore a locking of all the controls can be revoked directly at the unit at any time.

The new Digital Patch Mode will be supported (configuration and saving of setups) soon in an updated version of MIDI Remote. The Digital Patch Mode is already now remote controllable, see the table of MIDI commands in chapter 19.3 of this manual.

The software MIDI Remote also controls RME's ADI-648, ADI-6432, ADI-642, Micstasy, OctaMic XTC, the complete M-series and the MADI Bridge.





User's Guide



ADI-8 QS

▶ Inputs and Outputs

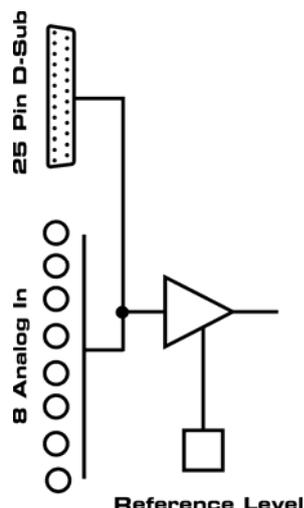
11. Analog Inputs / Outputs

11.1 Line In

The ADI-8 QS rear has 8 balanced Line inputs, provided by 1/4" TRS and a 25-pin D-sub connector. Both are internally connected, so can not be used at the same time. The electronic input stage is built in a servo balanced design which handles unbalanced (mono jacks) and balanced (XLR, stereo jacks) signals correctly, automatically adjusting the level reference.

When using unbalanced cables with XLR or TRS connectors be sure to connect pin 3 (- or ring) to 1 (ground). Otherwise noise may occur, caused by the unconnected negative input of the balanced input.

The 25-pin D-sub connector follows the pinout known from devices manufactured by Tascam (pinout see chapter 16.9). Your local dealer will supply analog breakout multicores D-sub to XLR in Tascam pinout in different lengths.



The ADI-8 QS uses a completely symmetrical signal path (including limiter) that guarantees an exceptional sound quality, outstanding low distortion and highest Signal to Noise ratio in all level settings.

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 ADI-8 QS internally uses hi-quality electronic switches, which introduce no additional noise or distortion to the audio path. The key INPUT allows for a perfect adaptation of all 8 channels to the most often used studio levels.

Each analog input has its own 7-segment level meter, so every channel can be checked for input signal and overloads easily. The level data incorporate the current Input Trim setting and the digital limiter. The red LED OVR lights up 0.2 dB before full scale level (-0.2 dBFS).

The ADI-8 QS uses the following level references:

Reference	0 dBFS @	Headroom @ +4 dBu	Other RME devices
+24	+24 dBu	20 dB	-
+19	+19 dBu	15 dB	LoGain
+13	+13 dBu	9 dB	+4 dBu
+4.2	+4.2 dBu	12 dB at -10 dBV	-10 dBV

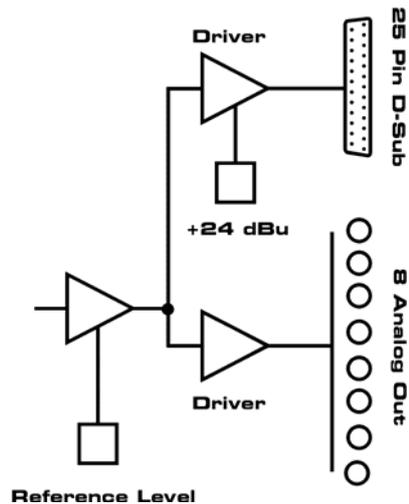
The setting +4.2 dBu equals -10 dBV with 12 to 15 dB headroom. Set to +24 dBu the ADI-8 QS is compatible to SMPTE (+24 dBu @ 0 dBFS, +4 dBu with 20 dB of headroom).

11.2 Line Out

The ADI-8 QS has 8 balanced Line outputs on the rear, provided by 1/4" TRS and a 25-pin D-sub connector. They have their own output drivers respectively and can therefore – in contrary to the inputs – be used simultaneously.

The electronic output stage of the TRS jacks is built in a servo-balanced design which handles unbalanced (mono plugs) and balanced (stereo plugs) correctly, automatically adjusting the level reference. The maximum output level is +21 dBu.

When selecting +24 dBu the LEDs +19 dBu and +24 dBu are lit at the same time, because the TRS jacks stay at the +19 setting, while the D-sub connector outputs are operated in the +24 setting.



The electronic output stage of the D-sub connector does not operate servo-balanced! When connecting unbalanced equipment, make sure pin 3 of the XLR output is not connected. A connection to ground can cause a higher THD value (higher distortion)!

The 25-pin D-sub connector follows the pinout known from devices manufactured by Tascam (pinout see chapter 16.9). Your local dealer will supply analog breakout multicores D-sub to XLR in Tascam pinout in different lengths. The maximum output level at the D-sub connector is +27 dBu.

Each output channel has its own 7-segment level meter, so every channel can be checked for output signal and overloads easily. The level data are taken directly in front of the DA-converter, so behind the digital level settings of Output Trim and Global Output Level. The red LED OVR lights up 0.2 dB before full scale level (-0.2 dBFS).

To maintain an optimum level for devices connected to the analog outputs, the ADI-8 QS internally uses hi-quality electronic switches, which introduce no additional noise or distortion to the audio path. The key OUTPUT allows for a perfect adaptation of all 8 channels to the most often used studio levels. The different output levels guarantee optimal conversion results, while still being compatible to any attached analog equipment.

The ADI-8 QS uses the following level references:

Reference	0 dBFS @	Headroom @ +4 dBu	Other RME devices
+24	+24 dBu	20 dB	-
+19	+19 dBu	15 dB	HiGain
+13	+13 dBu	9 dB	+4 dBu
+4.2	+4.2 dBu	12 dB at -10 dBV	-10 dBV

The setting +4.2 dBu equals -10 dBV with 12 to 15 dB headroom. Set to +24 dBu the ADI-8 QS is compatible to SMPTE (+24 dBu @ 0 dBFS, +4 dBu with 20 dB of headroom).

12. Digital Inputs / Outputs

12.1 AES/EBU

The four AES/EBU inputs and outputs are provided on the rear of the ADI-8 QS via a 25-pin D-sub connector with Tascam pinout (also used by Digidesign). A digital breakout cable will provide 4 male and 4 female XLR connectors. Every input and output is transformer-balanced and galvanically isolated.

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

In standard operation the AES outputs carry the converted analog input signal. When activating the option *Digital to Digital* in the Setup menu the currently chosen digital input data is present at the outputs instead, see chapter 9.15.

The ADI-8 QS supports Single Wire only, in the range of 32 kHz up to 192 kHz: a total of 8 channels, 2 channels per AES wire. The effective sample frequency equals the clock on the AES wire. In case a conversion from/to Single, Double and Quad Wire is required, the RME ADI-192 DD, an 8-channel universal sample rate and format converter, is highly recommended.

Besides the audio data, digital signals in SPDIF or AES/EBU format contain a channel status coding, which is being used for transmitting further information. The output signal coding of the ADI-8 QS has been implemented according to AES3-1992 Amendment 4:

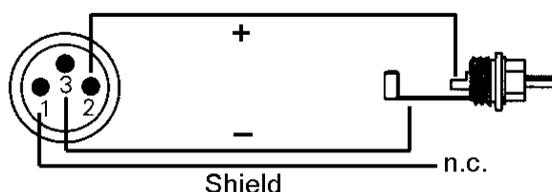
- 32 kHz, 44.1 kHz, 48 kHz, 64 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz according to the current sample rate
- Audio use
- No Copyright, Copy permitted
- Format Professional
- Category General, Generation not indicated
- 2-Channel, No Emphasis
- Aux bits audio use, 24 bit
- Origin: ADI8

Emphasis

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

 *An Emphasis indication gets lost! This information is neither passed on to the MAD1 output, nor to any of the AES outputs!*

Connecting devices with coaxial SPDIF ports to the ADI-8 QS inputs is accomplished by simple cable adapters XLR/RCA. To achieve this, pins 2 and 3 of an XLR plug are being connected to the two contacts of a Phono/RCA plug. The ground shield of the cable is only connected to pin 1 of the XLR plug.



The same is true for the outputs, but a male XLR connector is used then. Note that most consumer HiFi equipment with phono SPDIF inputs will only accept signals with Channel Status 'Consumer'! In such cases the above adapter cable will not work.

12.2 ADAT Optical

The ADI-8 QS provides two digital inputs and outputs in ADAT optical format. Using sample rates up to 48 kHz only the port labeled MAIN is relevant for operation. Higher sample rates than 48 kHz are realized by sample multiplexing. When using more than four channels at Double Speed or two channels at Quad Speed, the port labeled AUX has to be used as well.

The ADAT optical inputs of the ADI-8 QS use RME's unsurpassed Bitclock PLL, which 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.

In Single Speed mode both outputs carry the same audio data. Therefore the output signal can be sent to two devices at the same time.

In standard operation the ADAT outputs carry the converted analog input signal. When activating the option *Digital to Digital* in the Setup menu the currently chosen digital input data is present at the outputs instead, see chapter 9.15.

The ADAT outputs can be used in parallel to the AES outputs at up to 192 kHz, but in QS mode only channels 1 to 4 will be available.

ADAT MAIN

Interface for the first or only device sending/receiving an ADAT signal. Carries the channels 1 to 8. In S/MUX mode (Double Speed), this port carries the channels 1 to 4. In S/MUX4 mode (Quad Speed) this port carries channels 1 and 2.

ADAT AUX

Additional port for a transmission of channels 5 to 8 in S/MUX mode, or 3 and 4 in S/MUX4 mode.

With signals encoded as S/MUX, the Clock Section (STATE) has to be switched into DS mode manually, depending on the current application. Each port carries the data of 4 channels only, for up to 8 channels MAIN *and* AUX have to be used.

With signals encoded as S/MUX4, the Clock Section (STATE) has to be switched into QS mode manually, depending on the current application. Each port carries the data of 2 channels only, for up to 4 channels MAIN *and* AUX have to be used.

12.3 I64 MADi Card

The I64 MADi Card provides the ADI-8 QS with a 64-channel MADi input and output. Coaxial and optical outputs operate simultaneously and deliver the same data. The card's ID determines which MADi channels the ADI-8 QS will use (see chapter 9.5, ID). The MADi outputs operate in parallel to the AES/EBU and ADAT outputs, provide the same audio data, and are configured with the same front panel controls.

The I64 MADi Card also features optical and coaxial MADi input. The input is switched automatically, according to where a valid input signal is detected. This ensures full redundancy operation, immediately changing to the other input in case of loss of the input signal.

Since late 2011 a newer version of the i64 MADi card is shipped, having two green Sync LEDs beside the inputs. Here the above automatic input switching can be changed to different modes, like giving optical or coaxial input priority. The leaflet included with the i64 MADi card explains those options in detail. This newer version also enables the special Digital Patch Mode, see chapter 9.16.



The MADi input will operate as an optional clock source (Clock section, OPTN) as well as a through input. Since each ADI-8 QS uses only 8 channels, up to 56 channels can be passed through.

This technique is used to serially cascade several ADI-8 QS. Incoming MADi data is passed through unchanged, only one block of eight channels is replaced, according to the ADI-8 QS' ID. This allows up to 8 devices to be connected serially. All 64 combined channels are available at the last device's output. The ID determining the block of eight channels used by an individual device is determined automatically (Auto ID) or manually in the setup menu (ID):

ID 01: Channels 1-8	ID 02: Channels 9-16	ID 03: Channels 17-24
ID 04: Channels 25-32	ID 05: Channels 33-40	ID 06: Channels 41-48
ID 07: Channels 49-56	ID 08: Channels 57-64	

The I64 MADi Card defaults to the 56-channel format at its outputs. As soon as a 64-channel format is detected at the inputs, or the ID 08 is used, the output switches to 64-channel format.

Note: The 96k frame mode is activated at the outputs automatically when the MADi input signal is 96k frame. A manual selection of the output format, and with this a converter functionality 48k/96k, is not available.

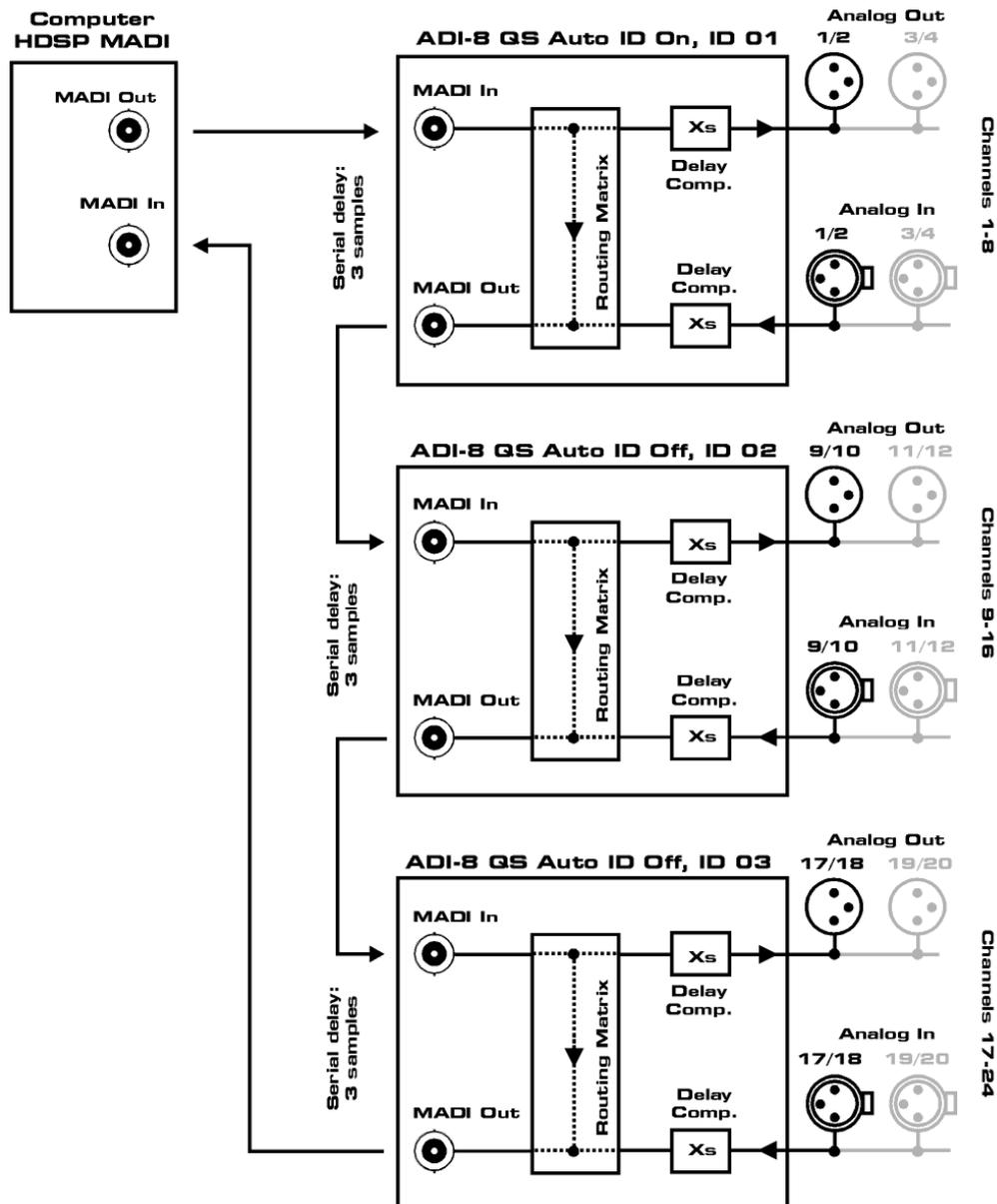
With an installed I64 MADi Card the ADI-8 QS can be remote controlled via MADi. At the same time MIDI data are transmitted via MADi, see chapter 10.3.

When multiple devices are connected serially, the MADi I/O of each ADI-8 QS causes a delay of 3 samples. Therefore at the MADi output of the last device, the data of all previous devices are delayed. At Double Speed the delay rises to 6 samples per unit, at Quad Speed to 12 samples.

The problem of this offset is solved by the function *Delay Compensation*, see chapter 9.8. It delays the signals in a way that they are sample-synchronous in multi-device operation. The diagram on the next page shows a serial setup with HDSP MADi card, three ADI-8 QS and activated Delay Compensation.



Delay Compensation has to be manually activated in each unit!



12.4 Differences serial MADI with I64 MADI Card and ADI-642

I64 MADI Card: Activate Auto ID in the first unit (ID of the master can be changed). All following units become Slave, get a consecutive ID, and with this the according channel routing. If desired, Delay Compensation has to be manually activated in each unit.

ADI-642: Activate ADC (Auto Delay Compensation) in the first unit (ID of the master is always 1). All following units become Slave, and compensate the respective delay automatically. The routing is done according to the setting of the matrix. If *Auto* (Auto Channel Assignment) is activated in the first unit too, all following units get a consecutive ID, and with this the according channel routing.

Mixed setups: Auto ID and Auto are compatible. In contrast (A)DC is automatic only with the 642. The ADI-8 QS require to activate Delay Compensation manually in each unit.

13. Word Clock

13.1 Word Clock Input and Output

Input

The ADI-8 QS transformer isolated word clock input is active when WCK is chosen in the clock section. The signal at the BNC input can be Single, Double or Quad Speed, the ADI-8 QS automatically adapts to it. As soon as a valid signal is detected, the WCK LED is constantly lit, otherwise it is flashing.

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 (0.3 V) input level are sufficient in principle. An additional hysteresis reduces sensitivity to 1.0 V, so that over- and undershoots and high frequency disturbances don't cause a wrong trigger.

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



Output

The ADI-8 QS word clock output is constantly active, providing the current sample frequency as word clock signal. In master mode, the word clock will be fixed to 44.1 kHz or 48 kHz (DS x 2, QS x 4). In any other case the sample rate is identical to the one present at the currently chosen clock input. When the current word clock source fails, the last valid sample rate will be held automatically.

Selecting the option *Si* in the Setup menu *Co* causes the word clock output 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 word clock signal received by the ADI-8 QS can be distributed to other devices by using the word clock output. With this the usual T-adaptor can be avoided, and the ADI-8 QS 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 ADI-8 QS 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.

13.2 Operation and Technical Background

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

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

In a digital studio synchronization 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 ADI-8 QS uses its internal clock, all other devices must be set to 'Slave' mode.

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 much 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 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 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.

The actual end of these problems is offered by the **SteadyClock** technology of the ADI-8 QS. Combining the advantages of modern and fastest digital technology with analog filter techniques, re-gaining a low jitter clock signal of 22 MHz from a slow word clock of 44.1 kHz is no problem anymore. Additionally, jitter on the input signal is highly rejected, so that even in real world usage the re-gained clock signal is of highest quality.

13.3 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. Actually you will find all the necessary components (T-adapters, terminators, cables) in most electronics and computer stores. The latter usually carries 50 Ohm components. The 75 Ohm components used for word clock are part of video technology (RG59).

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 as low impedance types, but all word clock inputs as high impedance types, 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 larger studios.

The ADI-8 QS word clock input can be high-impedance or terminated internally, ensuring maximum flexibility. If termination is necessary (e.g. because the ADI-8 QS is the last device in the chain), push the switch at the back (see chapter 13.1).

In case the ADI-8 QS 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 ADI-8 QS, we recommend to not pass the input signal via T-adapter, but to use the ADI-8 QS 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 held at the last valid frequency.

14. MIDI

The ADI-8 QS has a standard MIDI input and output, a 5-pin DIN jack each. The MIDI I/O is used for:

- remote control of the ADI-8 QS, see chapter 10.2
- transmission of MIDI data and remote control commands over MADI, in case the optional I64 MADI Card has been fitted, see chapter 10.3.



15. Remote

This mini-DIN jack can only be used to connect the included remote control. The 5 meter (16 ft) long cable is fixed to the Hardware Remote unit. More information on usage and functionality is found in chapter 10.1.



Note: The Hardware Remote uses the function Global Output Level, see chapter 9.2. The volume can not be changed if this option is deactivated in the Setup menu.



User's Guide



ADI-8 QS

▶ **Technical Reference**

16. Technical Specifications

16.1 Analog

Line In 1-8, TRS/D-Sub

- Input: 6.3 mm TRS jack and D-sub 25-pin, servo-balanced
- Input impedance: 10 kOhm
- Input sensitivity switchable +24 dBu, +19 dBu, +13 dBu, +4.2 dBu @ 0 dBFS

Analog Limiter

- Maximum analog input level unclipped: +30 dBu
- Threshold On: -3 dBFS
- Threshold Off: +3 dB
- THD+N: 0.03%, -52 dB
- Attack time: 3 ms
- Release time: 2-step, digitally controlled

AD-Conversion

- Resolution: 24 bit
- Signal to Noise ratio (SNR) @ +24 dBu, 44.1 kHz: 114.0 dB RMS unweighted, 117 dBA
- Signal to Noise ratio (SNR) @ +19 dBu: 114.0 dB RMS unweighted, 117 dBA
- Signal to Noise ratio (SNR) @ +13 dBu: 113.6 dB RMS unweighted, 116.6 dBA
- Signal to Noise ratio (SNR) @ +4.2 dBu: 109.8 dB RMS unweighted, 113 dBA
- Frequency response @ 44.1 kHz, -0.5 dB: 5 Hz – 21.5 kHz
- Frequency response @ 96 kHz, -0.5 dB: 5 Hz – 45.5 kHz
- Frequency response @ 192 kHz, -1 dB: 5 Hz – 66.5 kHz
- THD: < -110 dB, < 0.00032 %
- THD+N: < -104 dB, < 0.00063 %
- Channel separation: > 110 dB

DA-Conversion

- Resolution: 24 bit
- Signal to Noise ratio (SNR) @ +24 dBu, 44.1 kHz: 117,0 dB RMS unweighted, 120 dBA
- Signal to Noise ratio (SNR) @ +19 dBu: 117 dB RMS unweighted, 120 dBA
- Signal to Noise ratio (SNR) @ +13 dBu: 117 dB RMS unweighted, 120 dBA
- Signal to Noise ratio (SNR) @ +4.2 dBu: 114 dB RMS unweighted, 117 dBA
- Frequency response @ 44.1 kHz, -0.5 dB: 5 Hz – 22 kHz
- Frequency response @ 96 kHz, -0.5 dB: 5 Hz – 45.9 kHz
- Frequency response @ 192 kHz, -1 dB: 5 Hz - 90 kHz
- THD: < -104 dB, < 0.00063 %
- THD+N: < -102 dB, < 0.0008 %
- Channel separation: > 110 dB

Line Out 1-8, TRS

- Maximum output level: +21 dBu
- Output: 6.3 mm TRS jack, servo-balanced
- Output impedance: 75 Ohm
- Output level switchable +4.2 dBu, +13 dBu, +19 dBu @ 0 dBFS

Line Out 1-8, D-Sub

- Maximum output level: +27 dBu
- Output: D-sub 25-pin, electronically balanced
- Output impedance: 150 Ohm
- Output level switchable +4.2 dBu, +13 dBu, +19 dBu, +24 dBu @ 0 dBFS

16.2 Digital Inputs

AES/EBU

- 4 x 25-pin D-sub, transformer-balanced, galvanically isolated, according to AES3-1992
- High-sensitivity input stage (< 0.3 Vpp)
- SPDIF compatible (IEC 60958)
- Accepts Consumer and Professional format
- Lock Range: 27 kHz – 200 kHz
- Jitter when synced to input signal: < 1 ns
- Jitter suppression: > 30 dB (2.4 kHz)

ADAT Optical

- 2 x TOSLINK, according to Alesis specification
- Standard: 8 channels 24 bit, up to 48 kHz
- S/MUX: 2 x 8 channels 24 bit / 48 kHz, equalling 8 channels 24 bit 96 kHz
- S/MUX4: 2 x 8 channels 24 bit / 48 kHz, equalling 4 channels 24 bit 192 kHz
- Bitclock PLL ensures perfect synchronization even in varispeed operation
- Lock range: 31.5 kHz – 54 kHz
- Jitter when synced to input signal: < 1 ns
- Jitter suppression: > 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 Vpp – 5.6 Vpp
- Lock Range: 27 kHz – 200 kHz
- Jitter when synced to input signal: < 1 ns
- Jitter suppression: > 30 dB (2.4 kHz)

I64 MADI Card

- Coaxial via BNC, 75 Ohm, according to AES10-1991
- High-sensitivity input stage (< 0.2 Vpp)
- Optical via FDDI duplex SC connector
- 62.5/125 and 50/125 compatible
- Accepts 56 channel and 64 channel mode, and 96k frame
- Single Wire: up to 64 channels 24 bit 48 kHz
- Double Wire / 96k frame: up to 32 channels 24 bit 96 kHz
- Quad Wire: up to 16 channels 24 bit 192 kHz
- Lock range: 28 kHz – 54 kHz
- Jitter when synced to input signal: < 1 ns
- Jitter suppression: > 30 dB (2.4 kHz)

16.3 Digital Outputs

AES/EBU

- 4 x, transformer-balanced, galvanically isolated, according to AES3-1992
- Output voltage Professional 4.0 Vpp
- Format Professional according to AES3-1992 Amendment 4
- Single Wire: 4 x 2 channels 24 bit, up to 192 kHz

ADAT

- 2 x TOSLINK
- Standard: 8 channels 24 bit, up to 48 kHz
- S/MUX: 16 channels 24 bit / 48 kHz, equalling 8 channels 24 bit 96 kHz
- S/MUX4: 16 channels 24 bit / 48 kHz, equalling 4 channels 24 bit 192 kHz

Word Clock

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

I64 MADI Card

- Coaxial via BNC, 75 Ohm, according to AES10-1991
- Output voltage 600 mVpp
- Cable length coaxial: up to 100 m
- Optical via FDDI duplex SC connector
- 62.5/125 and 50/125 compatible
- Cable length optical: more than 500 m
- Generates 56 channel and 64 channel mode, and 96k frame
- Single Wire: up to 64 channels 24 bit 48 kHz
- Double Wire / 96k frame: up to 32 channels 24 bit 96 kHz
- Quad Wire: up to 16 channels 24 bit 192 kHz

16.4 Digital

- Clocks: Internal, AES In, ADAT In, Word Clock In, Option In
- Low Jitter Design: < 1 ns in PLL mode, all inputs
- Internal clock: 800 ps Jitter, Random Spread Spectrum
- Jitter suppression of external clocks: > 30 dB (2.4 kHz)
- Effective clock jitter influence on AD-conversion: near zero
- PLL ensures zero dropout, even at more than 100 ns jitter
- Supported sample rates: 28 kHz up to 200 kHz

16.5 MIDI

- 16 channels MIDI
- 5-pin DIN jacks
- Optocoupled, ground-free input

I64 MADI Card

- Invisible transmission via User bit of channel 56 (48k frame)

16.6 General

- Power supply: Internal switching PSU, 100 - 240 V AC, 40 Watts
- Typical power consumption: 23 Watts
- Maximum power consumption: < 30 Watts
- Dimensions including rack ears (WxHxD): 483 x 44 x 242 mm (19" x 1.73" x 9.5")
- Dimensions without rack ears/handles (WxHxD): 436 x 44 x 235 mm (17.2" x 1.73" x 9.3")
- Weight: 3 kg (6.6 lbs)
- Temperature range: +5° up to +50° Celsius (41° F up to 122°F)
- Relative humidity: < 75%, non condensing

16.7 Firmware

The ADI-8 QS is internally based on programmable logic. By re-programming of a little component called Flash-PROM, both function and behaviour of the unit can be changed at any time.

At the time of writing this manual, the unit is shipped with firmware 1.2. The firmware version is displayed after power on for about one second on the level meters of the ADI-8 QS.

Firmware 1.1: Initial release

Firmware 1.2: Seldom problem with the combination of word clock and AES input fixed.

Firmware 3.1: Enables the Digital Ptach Mode, when an i64 MADi card version 2 (green sync LEDs) is present in the unit.

16.8 MADi User Bit Chart

- RS-232: channels 1 to 9 (through mode active in the I64 MADi Card)
- ADC: channel 19
- MIDI: channel 56 (48k) / 28 (96k)

16.9 Connector Pinouts

D-Sub AES/EBU

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

Tascam / Digidesign:

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

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

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

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

Yamaha:

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

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

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

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

Euphonix:

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

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

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

D-Sub Analog Inputs / Outputs

The 25 pin D-sub connectors of analog input and output are wired according to the Tascam scheme, as shown in this table:

Channel	1+	1-	2+	2-	3+	3-	4+	4-	5+	5-	6+	6-	7+	7-	8+	8-
D-sub	24	12	10	23	21	9	7	20	18	6	4	17	15	3	1	14

GND is connected to pins 2, 5, 8, 11, 16, 19, 22, 25. Pin 13 is unconnected.

The servo balanced input circuitry allows to use unbalanced connections with no loss in level. For this to work, pins 3 (-) and 1 (GND) have to be connected.

The output circuitry does not operate in a servo-balanced way. When connecting unbalanced devices make sure that pin 3 (-) stays unconnected.

TRS jacks analog inputs / outputs

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

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

The servo-balanced 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.

17. Technical Background

17.1 Terminology

Single Speed

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

Double Speed

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

Quad Speed

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

Single Wire

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

Double Wire

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

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

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

Quad Wire

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

S/MUX

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

S/MUX4

The Quad Wire method realizes the transmission of two channels at up to 192 kHz via ADAT. The method is referred to as S/MUX4.

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

17.2 Lock and SyncCheck

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

If an AES or MADI signal is applied to the ADI-8 QS, the corresponding LED starts flashing. The unit indicates LOCK, i. e. a valid input signal (in case the signal is 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 ADI-8 QS is set to 44.1 kHz internal clock (clock mode master), and a mixing desk with MADI output is connected to the I64 MADI input. The OPTN LED will start flashing immediately, because the mixing desk's sample rate is generated internally, and thus slightly higher or lower than the ADI-8 QS'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 ADI-8 QS from internal clock to MADI (its internal clock will then be the clock delivered by the mixing desk). But in case another asynchronous 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 ADI-8 QS includes **SyncCheck**[®]. It checks all clocks used for *synchronicity*. If they are not synchronous to each other (i. e. absolutely identical), the LED of the asynchronous input flashes. In case they are synchronous the LED stays dark, only the LED of the current clock source will be lit (constantly). In example 1 it would have been obvious that the OPTN LED starts flashing after connecting the mixing desk.

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

17.3 Latency and Monitoring

The term **Zero Latency Monitoring** has been introduced by RME in 1998 for the DIGI96 series and describes 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*, found on 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. RME's digital receiver's buffer and the output via the transmitter cause 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.

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 about 40 samples, equalling one millisecond. A playback and re-record of the same signal via DA and AD (loop-back) then causes an offset of the newly recorded track of about 2 ms.

Low Latency!

The ADI-8 QS uses high-class AD- and DA-converters from Cirrus Logic, offering exceptional Signal to Noise and distortion figures. But the biggest difference to all other converter chips is their innovative digital filter, achieving for the first time a delay of only 12 samples in Single Speed, 9 samples in Double Speed, and 5 (!) samples in Quad Speed. The exact delays caused by the AD- and DA-conversion of the ADI-8 QS are:

Sample frequency kHz	44.1	48	88.2	96	176.4	192
AD (12 x 1/fs) ms	0.27	0.25				
AD (9 x 1/fs) ms			0.1	0.09		
AD (5 x 1/fs) ms					0.028	0.026
DA (10 x 1/fs) ms	0.22	0.2				
DA (5 x 1/fs) ms			0.056	0.052		
DA (5 x 1/fs) ms					0.028	0.026

These values are less than a quarter of those available from even much more expensive devices. They represent an important step in further reducing the latency in the computer-based recording studio. At Double Speed and Quad Speed the added latency can simply be ignored. In short: the ADI-8 QS turns 'analog digital monitoring' into real analog-style monitoring.

17.4 DS - Double Speed

When activating the *Double Speed* mode the ADI-8 QS 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 (Sample Multiplexing)* in connection with the ADAT format.

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 ADI-8 QS automatically uses *Sample Multiplexing* 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 Port	1/2 ADAT1	3/4 ADAT1	5/6 ADAT1	7/8 ADAT1	1/2 ADAT2	3/4 ADAT2	5/6 ADAT2	7/8 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.

17.5 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 (S/MUX4) results in only two channels per optical output. Therefore in Quad Speed mode the ADI-8 QS is limited to 4 channels at the ADAT outputs.

The AES outputs provide 192 kHz as Single Wire only.

17.6 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	Unbalanced
Impedance	110 Ohm	75 Ohm
Level	0.2 V up to 5 Vpp	0.2 V up to 0.5 Vpp
Clock accuracy	not specified	I: ± 50 ppm II: 0.1% III: Variable Pitch
Jitter	< 0.025 UI (4.4 ns @ 44.1 kHz)	not specified

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

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

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

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

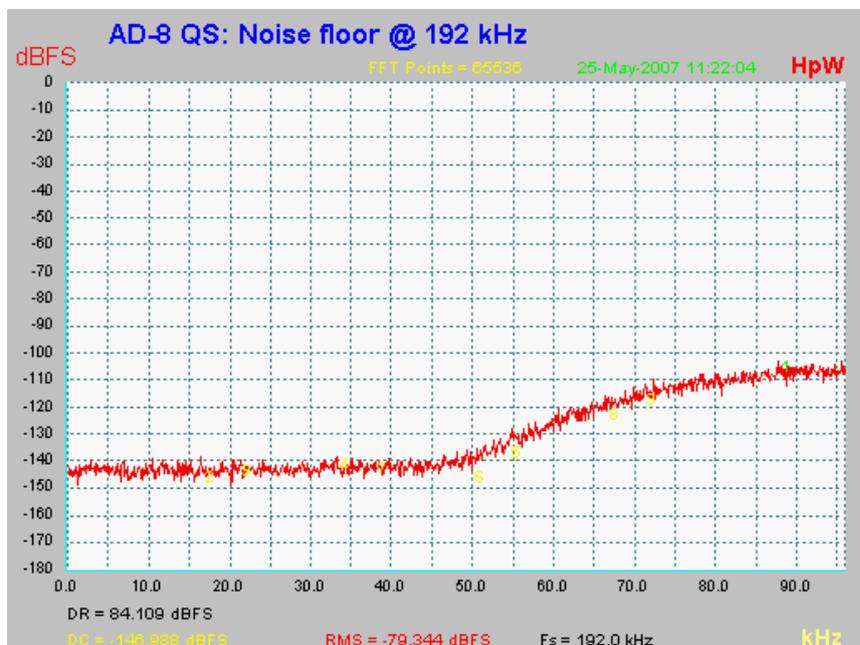
17.7 Signal to Noise Ratio in DS- / QS-Operation

The outstanding signal to noise ratio of the ADI-8 QS AD-converters can be verified even without expensive test equipment, by using record level meters of various software. But when activating the DS and QS mode, the displayed noise level will rise from -114 dBFS to -110 dBFS at 96 kHz, and -79 dBFS at 192 kHz. This is not a failure. The software measures the noise of the whole frequency range, at 96 kHz from 0 Hz to 48 kHz (RMS unweighted), at 192 kHz from 0 Hz to 96 kHz.

When limiting the measurement range from 20 Hz to 20 kHz (so called audio bandpass) the value would be -114 dB again. This can be verified with RME's *DIGICheck*. The function **Bit Statistic & Noise** measures the noise floor by *Limited Bandwidth*, ignoring DC and ultrasound.

Subframe	MSB	Audio Data	LSB	AUX	CUV	RMS LB [dB+3]	RMS [dBA+3]	DC [dB]	
1 - Left	XXXXXXXXXXXXXXXXXXXX	XXXXXXXXXXXXXXXXXXXX	XXXXXXXXXXXX	XXXXXXXXXXXX	000	-113.9	-116.5	-143.7	
2 - Right	XXXXXXXXXXXXXXXXXXXX	XXXXXXXXXXXXXXXXXXXX	XXXXXXXXXXXX	XXXXXXXXXXXX	000	-113.9	-116.5	-142.4	
Bits	4	8	12	16	20	24	20Hz ... 20kHz	A-weighting	0Hz

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 45 kHz. Therefore the noise is slightly increased in the ultrasound area. High-frequency noise has a high energy. Add the 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.



As can be seen in the above picture, the noise floor stays fully unchanged up to 45 kHz. With sample rates up to 96 kHz the noise shaping completely happens outside of the transmission range.

17.8 MADl Basics

MADl, the serial **M**ultichannel **A**udio **D**igital **I**nterface, has been defined already in 1989 as an extension of the existing AES3 standard following several manufacturers' wish. The format also known as AES/EBU, a balanced bi-phase signal, is limited to two channels. Simply put, MADl contains 28 of those AES/EBU signals in serial, i. e. after one another, and the sample rate can still even vary by +/-12.5%. The limit which cannot be exceeded is a data rate of 100Mbit/s.

Because an exact sampling frequency is used in most cases, the 64 channel mode was introduced officially in 2001. It allows for a maximum sample rate of 48 kHz + ca. 1%, corresponding to 32 channels at 96 kHz, without exceeding the maximum data rate of 100 Mbit/s. The effective data rate of the port is 125 Mbit/s due to additional coding.

Older devices understand and generate only the 56 channel format. Newer devices often work in the 64 channel format, but offer still no more than 56 audio channels. The rest is being eaten up by control commands for mixer settings etc.. The ADI-6432 shows that this can be done in a much better way, with an invisible transmission of 16 MIDI channels plus serial RS232 data stream, and the 64-channel MADl signal still being 100% compatible.

For the transmission of the MADl signal, proved methods known from network technology were applied. Most people know unbalanced (coaxial) cables with 75 Ohms BNC plugs, they are not expensive and easy to get. The optical interface is much more interesting due to its complete galvanic separation, but for many users it is a mystery, because very few have ever dealt with huge cabinets full of professional network technology. Therefore here are some explanations regarding 'MADl optical'.

- The cables used are standard in computer network technology. They are thus not at all expensive, but unfortunately not available in every computer store.
- The cables have an internal fibre of only 50 or 62.5 µm diameter and a coating of 125 µm. They are called network cables 62.5/125 or 50/125, the former mostly being blue and the latter mostly being orange. Although in many cases not clearly labelled, these are always (!) glass fibre cables. Plastic fibre cables (POF, plastic optical fibre) can not be manufactured in such small diameters.
- The plugs used are also an industry standard and called SC. Please don't mix them up with ST connectors, which look similar to BNC connectors and are being screwed. Plugs used in the past (MIC/R) were unnecessarily big and are not being used any longer.
- The cables are available as a duplex variant (2 cables being glued together) or as a simplex variant (1 cable). The ADI-6432's optical module supports both variants.
- The transmission uses the multimode technique which supports cable lengths of up to almost 2 km. Single mode allows for much longer distances, but it uses a completely different fibre (8 µm). Due to the wave-length of the light being used (1300 nm) the optical signal is invisible to the human eye.

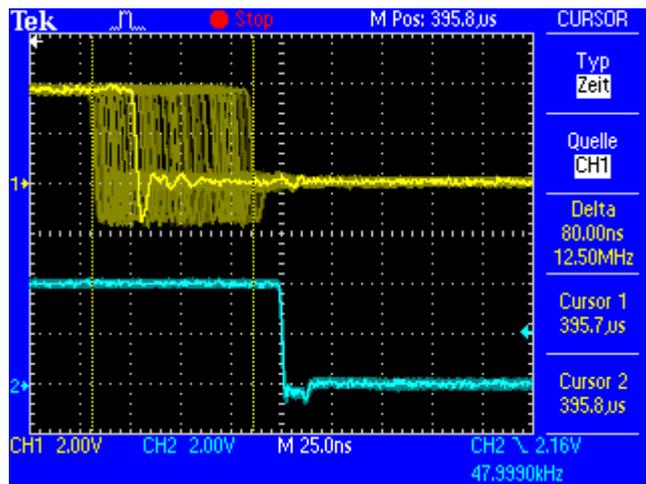
17.9 SteadyClock

The SteadyClock technology of the ADI-8 QS guarantees an excellent performance in all clock modes. Its highly efficient jitter suppression refreshes and cleans up any clock signal, and provides it as reference clock at the word clock output.

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

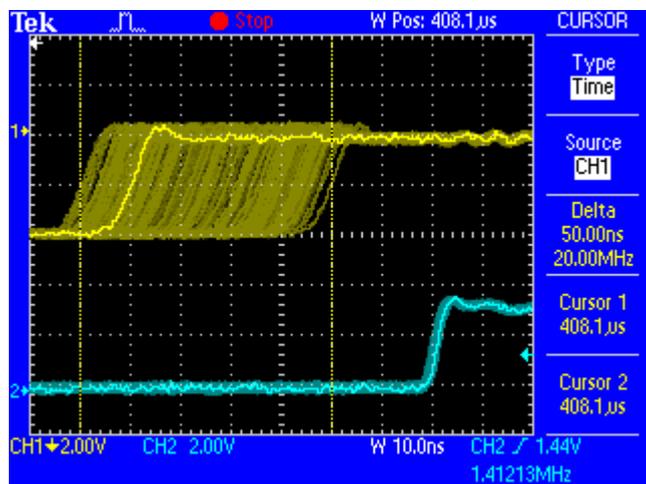
SteadyClock has originally been developed to gain a stable and clean clock from the heavily jittery MADI data signal. The embedded MADI clock suffers from about 80 ns jitter, caused by the time resolution of 125 MHz within the format. Common jitter values for other devices are 5 ns, while a very good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

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



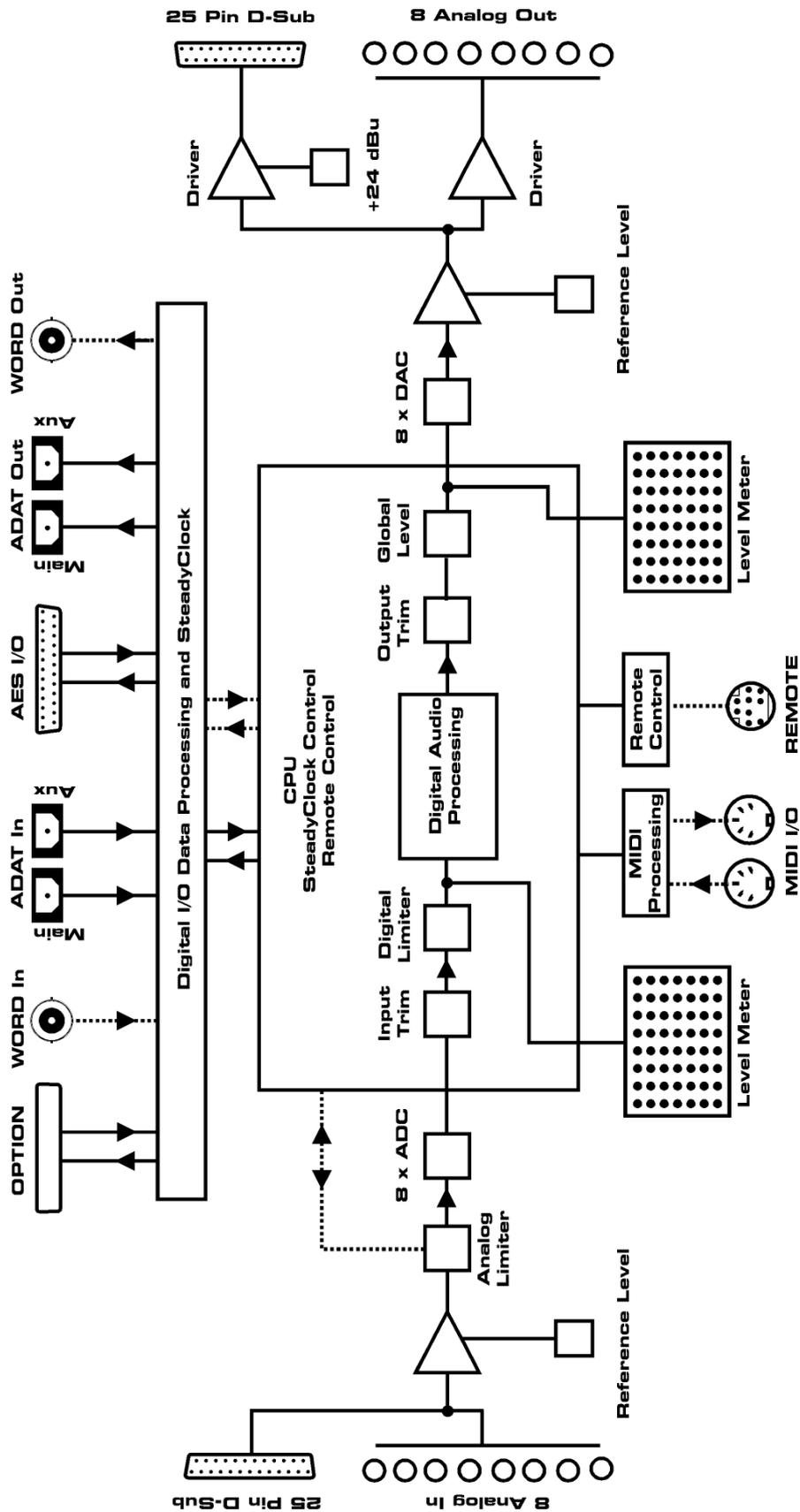
Using the other input sources of the ADI-8 QS, word clock, ADAT and AES/EBU, you'll most probably never experience such high jitter values. But SteadyClock is not only ready for these, it would also handle them just on the fly.

The screenshot to the right shows an extremely jittery word clock signal of about 50 ns jitter (top graph, yellow). Again SteadyClock provides an extreme clean-up. The filtered clock shows less than 2 ns jitter (lower graph, blue).



The cleaned and jitter-freed signal can be used as reference clock for any application, without any problem. The signal processed by SteadyClock is of course not only used internally, but also available at the ADI-8 QS' word clock output. It is also used to clock the digital outputs MADI, ADAT and AES/EBU.

18. Block Diagram



19. MIDI Implementation ADI-8 QS

19.1 Basic SysEx Format

<u>Value</u>	<u>Name</u>
F0h	SysEx header
00h 20h 0Dh	MIDITEMP manufacturer ID
41h	Model ID (ADI-8 QS)
00h..7Eh, 7Fh	Bank number / device ID (7Fh = broadcast, all IDs)
mm	Message type
nn	Parameter number (see table 1)
oo	Data byte
F7h	EOX

Bank Number / Device ID

The lower nibble refers to the device ID (0..7), the higher nibble refers to the bank number (0..7), e. g. 25h means bank 2, device 5. 7Fh addresses all banks and all devices.

19.2 Message Types

<u>Value</u>	<u>Name</u>
10h	Request value
11h	Request level meter data
20h	Set value
30h	Value response
31h	Level meter data response

Request Value

Format: F0 00 20 0D 41 (bank no. / dev ID) 10 F7

This string triggers a complete dump of all value response data bytes.

Request Level meter Data

Format: F0 00 20 0D 41 (Bank no. / dev ID) 11 F7

This string triggers a shorter dump of only the level meter data.

Set Value

Sets any number of parameters.

nn / oo can be repeated freely.

Value Response

After being triggered by receiving a request value command, device sends a string of all value response data bytes. Message type is set to 30h.

Level Meter Response

After being triggered by receiving a request level meter data command, device sends a string of all level meter data bytes. Message type is set to 31h.

Level meter Data Response Format

F0 00 20 0D 41 (bank no. / dev ID) 31 (AD ch.1) (ch.2) (ch.3) (ch.4) (ch.5) (ch.6) (ch.7) (ch.8)
(DA ch.1) (ch.2) (ch.3) (ch.4) (ch.5) (ch.6) (ch.7) (ch.8) F7

The peak level value will be stored and transmitted with the next level meter data request, and the stored value will be reset.

19.3 Table

No.	No.	Name	Set Val.	Val. Resp.	Data bytes
00h	0	Output Level	x	x	0..48 (see output level table)
01h	1	Trim Output Ch. 1	x	x	4..28, 0.5dB steps, 4 = -6dB, 16 = 0dB, 28 = +6dB
02h	2	Trim Output Ch. 2	x	x	see above
03h	3	Trim Output Ch. 3	x	x	see above
04h	4	Trim Output Ch. 4	x	x	see above
05h	5	Trim Output Ch. 5	x	x	see above
06h	6	Trim Output Ch. 6	x	x	see above
07h	7	Trim Output Ch. 7	x	x	see above
08h	8	Trim Output Ch. 8	x	x	see above
09h	9	Trim Input Ch. 1	x	x	see above
0Ah	10	Trim Input Ch. 2	x	x	see above
0Bh	11	Trim Input Ch. 3	x	x	see above
0Ch	12	Trim Input Ch. 4	x	x	see above
0Dh	13	Trim Input Ch. 5	x	x	see above
0Eh	14	Trim Input Ch. 6	x	x	see above
0Fh	15	Trim Input Ch. 7	x	x	see above
10h	16	Trim Input Ch. 8	x	x	see above
11h	17	Setup 1	x	x	see setup 1 table
12h	18	Setup 2	x	x	see setup 2 table
13h	19	Setup 3	x	x	see setup 3 table
14h	20	Setup 4	x	x	see setup 4 table
15h	21	Lock / Sync Info 1		x	see lock / sync table 1
16h	22	Lock / Sync Info 2		x	see lock / sync table 2
17h	23	Lock / Sync Info 3		x	see lock / sync table 3
18h	24	Mute Output 1..4	x	x	bit 0..3: 0 = normal operation, 1 = mute
19h	25	Mute Output 5..8	x	x	bit 0..3: 0 = normal operation, 1 = mute
1Ah	26	Reference Level	x	x	0..48, see output level table
1Bh	27	Setup 5	x	x	see setup 5 table
1Ch	28	Option 1 source	x	x	0 = Option 1,
1Dh	29	Option 2 source	x	x	1 = Option 2,
1Eh	30	Option 3 source	x	x	2 = Option 3,
1Fh	31	Option 4 source	x	x	3 = Option 4,
20h	32	Option 5 source	x	x	4 = Option 5,
21h	33	Option 6 source	x	x	5 = Option 6,
22h	34	Option 7 source	x	x	6 = Option 7,
23h	35	Option 8 source	x	x	7 = Option 8,
24h	36	ADAT 1 source	x	x	8 = ADAT 1,
25h	37	ADAT 2 source	x	x	9 = ADAT 2,
26h	38	AES source	x	x	10 = AES,
27h	39	Analog source	x	x	11 = Analog, 12 = n.c.
28h	40	(reserved)	x	x	
29h	41	(reserved)	x	x	
2Ah	42	(reserved)	x	x	
2Bh	43	(reserved)	x	x	
2Ch	44	Set bank no./Dev. ID	x		

Output Level:	0	-∞,
	1..13	-98, -94, -90, -86, -82, -78, -74, -70, -66, -62, -58, -54, -50
	14..28	-48, -46, -44, -42, -40, -38, -36, -34, -32, -30, -28, -26, -24, -22, -20
	29..47	-19, -18, -17, -16, -15, -14, -13, -12, -11, -10, -9, -8, -7, -6, -5, -4, -3, -2, -1
	48	0dB

		Setup 1	MSB / 7		0
			6	MSB / 2	analog input: 0 = +4dBu, 1 = +13dBu, 2 = +19dBu
			5	1	analog input: 3 = +24dBu, 4 = dig. Input
			4	LSB / 0	(analog input)
			3	MSB / 1	analog output: 0 = +4dBu, 1 = +13dBu,
			2	LSB / 0	analog output: 2 = +19dBu, 3 = +24dBu
			1	MSB / 1	limiter mode: 0 = off, 1 = digital,
			LSB / 0	LSB / 0	limiter mode: 2 = analog, 3 = analog + digital

		Setup 2	MSB / 7		0
			6		follow clock: 0 = off, 1 = on
			5		clock out: 0 = Fs, 1 = always single speed
			4	MSB / 2	clock select: 0 = int. 44.1, 1 = int. 48, 2 = WCK
			3	1	clock select: 3 = Option, 4 = AES, 5 = ADAT
			2	LSB / 0	clock select
			1	MSB / 1	clock range: 0 = single speed, 1 = ds, 2 = qs
			LSB / 0	LSB / 0	clock range

		Setup 3	MSB / 7		0
			6		Auto-Device: 0 = off, 1 = on
			5		Delay Compensation: 0 = off, 1 = on
			4	MSB / 1	Peak Hold: 0 = off, 1 = auto reset,
			3	LSB / 0	Peak Hold: 2 = manual, 3 = reset (tx only)
			2	MSB / 1	Digital Input: 0 = Option, 1 = ADAT, 2 = AES
			1	LSB / 0	Digital Input
			LSB / 0		Lock Keys: 0 = unlock, 1 = lock

		Setup 4	MSB / 7		0
			6	MSB / 1	fx drive: 0..3 = 1..4
			5	LSB / 0	(fx drive)
			4		analog output dim: 0 = normal level, 1 = dim
			3		direct level: 0 = disable, 1 = enable
			2		enable trim: 0 = disable, 1 = enable
			1		enable global level: 0 = disable, 1 = enable
			LSB / 0		display mode: 0 = normal, 1 = auto gain reduction

		Setup 5	MSB / 7		0
			6		
			5		
			4		
			3		
			2		
			1		
			LSB / 0		dd: 0 = di, 1 = fx

		Lock / Sync 1	MSB / 7		0
			6		0
			5		0
			4		0
		(Request only)	3		WCK Sync: 0 = no sync, 1 = sync
		(Request only)	2		WCK Lock: 0 = unlock, 1 = lock
		(Request only)	1		Option Sync: 0 = no sync, 1 = sync
		(Request only)	LSB / 0		Option Lock: 0 = unlock, 1 = lock

		Lock / Sync 2	MSB / 7		0
			6		0
			5		ADAT 1 Sync: 0 = no sync, 1 = sync
			4		ADAT 1 Lock: 0 = unlock, 1 = lock
		(Request only)	3		ADAT 2 Sync: 0 = no sync, 1 = sync
		(Request only)	2		ADAT 2 Lock: 0 = unlock, 1 = lock
		(Request only)	1		AES 1 Sync: 0 = no sync, 1 = sync
		(Request only)	LSB / 0		AES 1 Lock: 0 = unlock, 1 = lock

		Lock / Sync 3	MSB / 7		0
			6		0
			5		AES 2 Sync: 0 = no sync, 1 = sync
			4		AES 2 Lock: 0 = unlock, 1 = lock
		(Request only)	3		AES 3 Sync: 0 = no sync, 1 = sync
		(Request only)	2		AES 3 Lock: 0 = unlock, 1 = lock
		(Request only)	1		AES 4 Sync: 0 = no sync, 1 = sync
		(Request only)	LSB / 0		AES 4 Lock: 0 = unlock, 1 = lock

Request Level Meter Data

		Level meter Data	MSB / 7		0
			6		0
		(A/D only)	5	MSB / 2	gain reduction: 0 = no gr, 1 = < 1dB, 2 = > 1dB
		(A/D only)	4	/ 1	gain reduction: 3 = > 3dB, 4 = > 6dB, 5 = > 9dB
		(A/D only)	3	LSB / 0	gain reduction: 6 = > 12dB, 7 = > 15dB
			2	MSB / 2	level: 0 = < -60dBFS, 1 = < -30dBFS, 2 = < -12dBFS
			1	/ 1	level: 3 = < -6dBFS, 4 = < -3dBFS, 5 = < -1dBFS
			LSB / 0	LSB / 0	level: 6 = < 0dBFS, 7 = over