

User's Guide



DMC-842

The Digital Microphone Interface

SyncAlign™

164 Option Slot™

MultiMode™

SteadyClock™

SyncCheck™

8-Channel Digital Microphone Interface with Line Outputs
8-Channel AES to Analog / ADAT Interface
AES/EBU Format and Sample Rate Conversion
Optional 64-Channel MAD1 Interface
24 Bit / 192 kHz Digital Audio
MIDI Remote Control

 **adat®** AES-3
AES-10
AES42

24 Bit Interface

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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



DMC-842

▶ General

1. Introduction

The DMC-842 is both an AES/EBU interface as well as a controller for digital microphones. The unique device allows for connection and control of up to 8 digital microphones, and converting their signals to ADAT, AES/EBU, analog and (optional) MADI. Eight switchable hi-end sample rate converters offer a flexible clocking and further usage options.

When developing the DMC-842, RME worked closely with the microphone manufacturers to secure maximum compatibility and best functionality. As a result the DMC-842 is the most flexible and most compatible AES42 interface available – a true milestone for the broad acceptance of the new digital microphone technology.

2. Package Contents

Please check that your DMC-842 package contains each of the following:

- DMC-842
- Power cord
- Manual
- RME Driver CD
- 1 optical cable (TOSLINK), 6.6 ft (2 m)

3. Brief Description and Characteristics

The DMC-842 is an 8-channel interface for AES42 digital microphones, with full remote controllability and optional MADI I/O. In a standard 19" box with 2 unit height the device offers numerous extraordinary features like MultiMode, Intelligent Clock Control (ICC), SyncCheck, Steady-Clock, MIDI over MADI, and remote control via AES, MADI and MIDI.

- 8 XLR AES/EBU inputs
- 8 balanced XLR line outputs
- 8-channel 24 bit sample rate conversion
- Fully compatible to AES42
- Digital Phantom Power switchable per input
- Supports Mode 1 and Mode 2
- Gain adjustable per channel, even in asynchronous Mode 1
- Future-proof by flash updates
- LED level meter with 13 LEDs per channel
- Noise suppression on power-on and power-off at the analog outputs
- M/S decoding/encoding
- Fully remote controllable
- Word clock input and output
- SyncCheck tests and reports the synchronization status of the clock signals
- SyncAlign guarantees sample aligned and never swapping channels
- MIDI I/O
- 4 x AES/EBU Out per D-sub, 8 channels @ 192 kHz
- 2 x ADAT Out, 8 channels @ 96 kHz
- Optional MADI I/O (164 MADI Card)

4. First Usage – Quick Start

4.1 Controls - Connectors - Displays

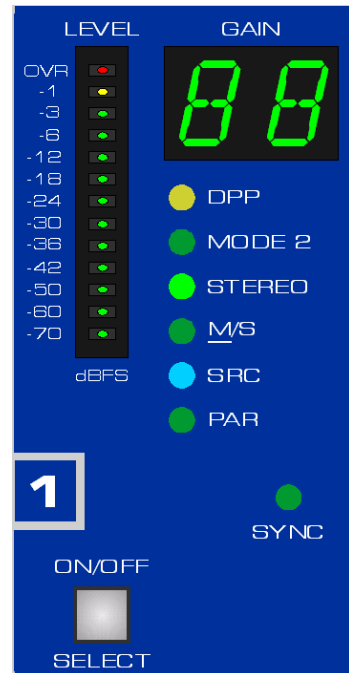
The front of the DMC-842 features eight LED level meter, eight numerical LED displays, eight Select keys, a rotary encoder, further keys for more specific configuration options, and 69 LEDs providing a detailed status display.

Each channel has a dedicated configuration area on the front plate. The display **GAIN** shows the currently set amount of amplification. The 13-segment **LEVEL** meter with switchable Peak Hold function shows the incoming digital level.

The **SELECT** button is used for a per-channel activation / deactivation of:

GAIN	Amount of amplification
DPP	Digital Phantom Power
STEREO	Stereo mode
M/S	Mid/Side encoding
SRC	Sample Rate Converter

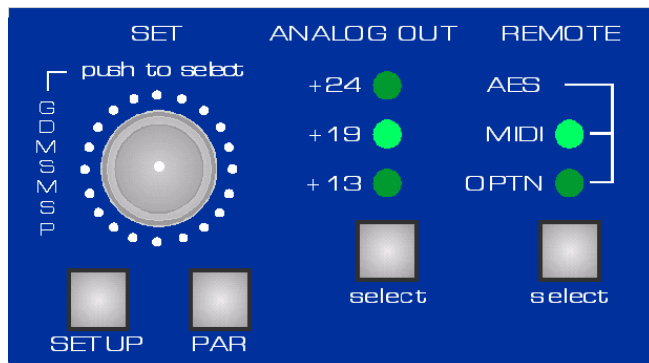
The LEDs **MODE 2** and **SYNC** are skipped, as these are status displays only. The also skipped LED **PAR** stands for the Parameter settings mode (see below).



Pushing the knob **SET** several times lets you access all functions consecutively. The rotary encoder is a highly intuitive multifunctional device. It is used to set the gain, to select the desired function, to deactivate the function of one or all channels, to select various options in the Setup menu and to perform the setting of the parameters.

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

REMOTE defines the source for MIDI remote control operation (Option Slot / MIDI, D-sub AES1, MIDI DIN jack, RS232).



After pressing the **SETUP** button the device enters the Setup menu, where many more options can be configured. More information can be found in chapter 10.

After pressing the **PAR** button the device enters the Parameter menu. More information can be found in chapter 11.

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

The rear panel of the DMC-842 has eight AES42 inputs, eight analog outputs, mains power, MIDI I/O, word clock I/O, the I64 Option Slot, and all digital inputs and outputs (AES/ADAT).

DIGITAL MICROPHONE – AES/EBU INPUTS (XLR): Eight balanced **AES42 inputs** with switchable digital phantom power (DPP). These inputs are compatible to **AES3** and **AES/EBU**.

ANALOG LINE BALANCED OUTPUTS (XLR): Eight balanced **line outputs** with up to +27 dBu level.

AES I/O (25-pin D-sub): The D-sub connector provides four AES/EBU outputs and one AES/EBU input (clock synchronization). The 25 pin D-sub connector is wired according to the widely spread Tascam standard (pinout see chapter 18.9). The AES I/Os are transformer-coupled.

ADAT OUT (TOSLINK): Optical ADAT outputs. These provide the same signals as the AES/EBU outputs, but in ADAT format.



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 DMC-842, and – with installed MADI Card - to transmit MIDI data via MADI.

IEC receptacle for power connection. The specially developed, internal hi-performance switch mode power supply lets the DMC-842 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 DMC-842 begins in the CLOCK section. Choose a clock source and a sample rate. The next step is the configuration of the input channels. When using digital microphones DPP (digital Phantom power) must be activated. In Mode 1 the SRCs have to be activated as well as soon as more than one microphone is connected.

The next step is the GAIN setting. This can be done in two ways:

- Individually: Hit the SELECT button of one or several channels. The corresponding GAIN displays start flashing. Select the desired value with the rotary encoder. Or:
- Globally: Push the encoder knob once. All GAIN displays start flashing. Select the desired value with the rotary encoder.

The flashing stops automatically after about six seconds. The LEVEL meter serves as comfortable tool to check whether gain is sufficient or already overloading the input.

To activate a function, press the SET knob repeatedly until the corresponding LEDs are flashing. Pressing once all GAIN displays start flashing (gain setting), pressing twice selects all DPP, then STEREO and so on. To activate or deactivate a specific function per channel, the corresponding **SELECT** button is used. They are therefore also labeled ON/OFF.

The DMC-842 stores all settings before switching off, and sets them automatically when switching on the next time. The storing process is triggered about 4 seconds after the last change. Using the included Windows software, the device's complete state can be stored and archived.

5. Accessories

RME offers several optional components for the DMC-842:

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)

164 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 DMC-842 undergoes comprehensive quality control and a complete test at IMM before shipping. The usage of high grade components should guarantee a long and trouble-free operation of the unit.

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 DMC-842. 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>

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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 RL2004/108/EG, and European Low Voltage Directive RL2006/95/EG.

FCC

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

RoHS

This product has been soldered lead-free and fulfils 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 DMC-842.

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.



User's Guide



DMC-842

▶ Usage and Operation

8. Front Panel Controls

8.1 Select Key and Rotary Encoder (SET)

The rotary encoder **SET** is a highly intuitive multifunctional device. It is used to set the gain, to select the desired function, to deactivate the function of one or all channels, to select various options in the Setup menu and to perform the setting of the parameters.

To activate a function, press the SET knob repeatedly until the corresponding LEDs are flashing. Pressing once all GAIN displays start flashing (gain setting), pressing twice selects all DPP, then STEREO and so on.

To activate or deactivate a specific function per channel, the corresponding **SELECT** button is used. They are therefore also labeled ON/OFF.

Each channel offers the following functions:

GAIN	Amount of amplification
DPP	Digital Phantom Power
MODE 2	Status display, not selectable
STEREO	Only odd channels: Stereo mode
INACTIVE	Only even channels: when Stereo mode had been activated
M/S	Mid/Side encoding
SRC	Sample Rate Converter
PAR	Parameter configuration, not selectable
SYNC	Sync state display, not selectable

The LEDs **MODE 2** and **SYNC** are skipped, as these are status displays only. The also skipped LED **PAR** stands for the Parameter settings mode (see chapter 11).

Setting the GAIN can be done in two ways:

- Individually: Hit the SELECT button of one or several channels. The corresponding GAIN displays start flashing. Select the desired value with the rotary encoder. Or:
- Globally: Push the encoder knob once. All GAIN displays start flashing. Select the desired value with the rotary encoder.

When adjusting several channels at the same time, the relation between individual channels will be retained. So the gain of multiple channels can be increased or decreased without them losing their level relations. In case one of the linked channels reaches the highest or lowest gain value (0 or +63 dB), the gain relationship gets lost.

Turning the encoder knob to the left initiates a global OFF command for all selected (flashing) channels. For example, DPP can be switched off for all channels by pressing the encoder knob twice and then turning it to the left by four ticks. The corresponding global ON command (a turn to the right by at least four ticks) is also implemented.

For safety reasons, the DMC-842 will leave the selected function after 6 seconds. This may seem cumbersome when changing the gain frequently, because the gain function will have to be selected every time via the Select button or the encoder first. On the other hand, the DMC-842 is absolutely safe from accidental changes.

8.2 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, AES, Option = MADI) and internal clock. The button SAMPLE RATE sets the sample rate for both external and 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 DMC-842. Therefore the external device has to be synchronized to the DMC-842's word clock output or AES/ADAT/MADI output.

The DMC-842 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 DMC-842's 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 DMC-842 should operate at 192 kHz, but receives a synchronous word clock of 48 kHz, the button SAMPLE RATE 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.

8.3 Analog Out

The button **ANALOG OUT** is used to set the analog reference level at the analog outputs. This level is reached at full scale of the DA-converters, thus matching the front panel level meters level indication.

Reference	0 dBFS @	Analog Headroom
+24	+24 dBu	3 dB
+19	+19 dBu	8 dB
+13	+13 dBu	14 dB

8.4 Remote

REMOTE defines the source of MIDI remote control commands. Choices are the MIDI DIN jack, the D-sub AES1 and the MADI input of the I64 MADI Card (Option Slot). A firmware update added a fourth option, RS232, signaled by the LEDs OPTN and MIDI flashing alternately. Nevertheless RS232 is currently not functional.

Note: Via MIDI remote control, all front panel controls 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.

9. The Input Channel in Detail

9.1 General

Each channel has a dedicated configuration area on the front plate. The display **GAIN** shows the currently set amount of amplification. The 13-segment **LEVEL** meter with switchable Peak Hold function shows the digital signal level after Gain control, but before SRC, stereo and M/S processing, as shown in the block diagram (chapter 20). The **SELECT** button is used to select a channel, but also to activate/deactivate the different functions (DPP, STEREO etc).

9.2 Gain

The DMC-842's **GAIN** can be set individually per channel over a range of 0 dB up to +63 dB, in steps of 1 dB. The device has its own gain control, operational even in the asynchronous Mode 1, and – as shown in the block diagram – is placed in front of the SRCs. When the microphone signals support for gain control, the DMC-842's gain stage is deactivated automatically by setting it to 0 dB internally.

The gain can not be set directly for inactive channels. Therefore in Stereo mode the gain display of the right channel is turned off.

9.3 Digital Phantom Power

The LED **DPP** indicates activated digital phantom power for the XLR input. Phantom power should only be activated when using digital microphones.



Connecting and disconnecting microphones while phantom power is active causes a voltage surge, which can destroy components. Switch phantom power off before connecting/disconnecting any external device.

As called for in the AES42 standard, the DMC-842 provides a common mode voltage of 10 Volts on both input pins. When connecting fully balanced AES/EBU outputs, this voltage theoretically stays without effect. Nevertheless we strongly recommend to switch off DPP in such cases. When connecting unbalanced outputs (AES-ID or SPDIF) a current is expected to flow through the output stage of the feeding device!

9.4 Mode 2

In Mode 1 the digital microphone is clock master, operating from the internal and fixed frequency. When connecting more than one microphone, the usage of sample rate converters for clock-decoupling is unavoidable.

In the newer Mode 2 the digital microphone is clock slave. The DMC-842 determines the frequency and generates a digital control signal for each microphone. A sample rate conversion is not required in this case.

The microphone signals the supported mode, the DMC-842 switches its input accordingly. The LED MODE 2 therefore is a state display only.

Note: Not every digital microphone supports Mode 2. With regard to the outstanding SRC built into the DMC-842 Mode 1 can be seen as equally good alternative. However the SRCs delay the audio signals by about 141 samples. At 44.1 kHz, this equals about 3 ms.

9.5 Stereo / Inactive

The DMC-842 has 8 single AES/EBU compatible inputs, being a 2-channel (stereo) input each. At the time of printing this manual, digital stereo microphones not yet exist. A digital microphone therefore provides only the information of one channel per input. For maximum flexibility the operation mode can be switched between 2-channel and 1-channel. The labeling of the input XLR connectors has been done to signal this flexibility. The odd ones are labeled Stereo (1/2), the even ones Mono (2).

When an odd channel is switched into **STEREO** mode, the according even channel's **INACTIVE** LED lights up automatically. The gain can not be set directly for inactive channels. Therefore in Stereo mode the gain display of the right channel is turned off.

9.6 M/S Processing

The DMC-842 includes a digital M/S processor. The channels have a fixed assignment: all odd ones are M, all even ones are S. The front panel shows an according labeling:

Channel 1 M/S (D)
Channel 2 M/S (D)

The mid/side principle is a special positioning technique for microphones, which results in a mid signal on one channel and a side signal on the other channel. The DMC-842's M/S processor transforms these information back into a standard stereo signal. The process sends the monaural mid channel to left and right, the side channel too, but phase inverted (180°) to the right channel. For a better understanding: the mid channel represents the function L+R, while the side channel represents L-R.

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

9.7 SRC

The DMC-842 includes eight individually switchable sample rate converters (SRC). The technical specifications indicate an exceptional conversion quality, formerly known only from synchronous SRC devices. The conversion in full 24 bit resolution operates practically without loss of signal quality, so no audible artifacts or noise are added.

The SRC of the DMC-842 offers a maximum conversion rate of 1:7 or 7:1, respectively. Thus, 192 kHz can be converted to any sample rate down to 32 kHz, and 32 kHz can be converted to any frequency up to 192 kHz.

When using the internal clock, every SRC also works as a jitter killer. However, the DMC-842 is equipped with SteadyClock, thus operating as perfect jitter killer with any clock source.

An SRC not only converts sample rates, it also serves as a clock decoupler. With SRC active, even non-synchronizable devices (CD-players, DAT machines, etc.) can be used in a setup of digital devices, just as if they were externally synchronized. The SRC decouples input and output clock, sets the output clock to the common reference, thus allowing the combination of different clock-sources without any clicks or dropouts. This is exactly what happens when using digital microphones in Mode 1.

9.8 PAR

After pressing the **PAR** button below the rotary encoder the DMC-842 enters the Parameter menu. Turning the encoder knob scrolls through all the available options. More information can be found in chapter 11.

9.9 Sync

The DMC-842 has 8 independent digital inputs. In order to avoid clicks and drop outs due to faulty or missing synchronicity, a special process called *SyncCheck* analyzes the incoming data.

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.

10. The Setup Menu

10.1 General

Some options and settings are accessed and changed very seldom. These have been collected in the Setup menu. The Setup menu is entered by pushing **SETUP**, situated below the encoder. *SEt* is displayed on the left part of the Gain displays. Turning the encoder scrolls through all the options, displayed as abbreviation in the right part of the Gain displays.

The Select key 8 allows to toggle the current setting, for example changing between On and Off. Otherwise the Select keys 7 and 8 can be used to increase/decrease the current value, for example the ID (1 to 8).

All changes are stored automatically. With a second push on the SETUP key the Setup menu is exit.

10.2 ID (Id)

Default: 01

Available settings: 01, 02, 03, 04, 05, 06, 07, 08

To remote control more than one DMC-842 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 also defines the 8-channel group within the MADI signal that is used to insert the device's audio data:

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	

When several DMC-842, Micstasy, ADI8-QS or ADI-642 units are connected via MADI, Auto ID normally takes care of the channel assignment (see chapter 10.4, 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 Out* the ID also defines which MADI input channels are sent to the ADAT/AES outputs, see chapter 10.9. With activated mode *Analog Out* the ID also defines which MADI input channels are sent to the analog outputs, see chapter 10.10.

Note: When the device is *Auto ID Slave*, the displayed ID can not be changed manually.

10.3 Bank (bA)

Default: 01

Available settings: 01, 02, 03, 04, 05, 06, 07, 08

To remote control more than one DMC-842 each device can have its own ID (chapter 10.2), providing a separated remote control of multiple devices via a single MIDI channel. Additionally, not only 8 IDs but 8 banks of 8 IDs each are available.

10.4 Auto ID (Au)

Default: Off

Available settings: On, Off

This option relates to the use of the I64 MADI Card. Multiple DMC-842 (and ADI-642/Micstasy/ADI-8 QS, see chapter 14.3/14.4) connected serially can assign consecutive IDs to themselves. For this to happen, Auto ID is set to *On* at the first device of the chain. All other devices automatically turn into slaves. An Auto ID slave has the middle dot lit in the GAIN display of channel 8.

10.5 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 DMC-842 causes a delay of 3 samples. Therefore at the MADI output of the last device, the data of all upstreamed 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.



Delay Compensation has to be manually activated 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 DMC-842 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 μ s.

18 samples @ 96 kHz
equal 187 μ s.

12 samples @ 192
kHz equal 62.5 μ s.

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.

10.6 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 DMC-842 can use it in any case, independent of its current clock range. When activating the option *Follow Clock* the DMC-842 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, as it is not possible to detect the sample rate range of the MADI-embedded audio data.

10.7 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 DMC-842 this limitation is also valid for the AES output, as the device internally handles all the data streams in S/MUX mode.

10.8 Peak Hold (Ph)

Default: Off

Available settings: On, Off

The level meter can be configured to hold the peak value permanently. A reset is done by turning the Encoder, even when no other function had been selected before.

10.9 Digital Output (do)

Default: 42

Available settings: 42, oP

This option relates to the use of the I64 MADI Card. In default state (42 = Mic) the XLR input signal is available at all three digital outputs. Selecting oP (Option) an eight-channel group of the MADI input signal is sent to the ADAT/AES outputs. With this the MADI connection can also be used as return path. Sending audio via MADI to the DMC-842, the audio becomes available at the DMC-842's digital outputs ADAT/AES. The MADI output continues to carry the XLR input signals and the fed-through input data.

Note: The 8-channel group sent to ADAT/AES is defined by the currently chosen ID.

10.10 Analog Output (Ao)

Default: 42

Available settings: 42, oP

This option relates to the use of the I64 MADI Card. In default state (42 = Mic) the XLR input signal is available at the analog outputs. Selecting oP (Option) an eight-channel group of the MADI input signal is sent to the analog outputs. With this the MADI connection can also be used as return path. Sending audio via MADI to the DMC-842, the audio becomes available at the DMC-842's analog outputs. The MADI output continues to carry the XLR input signals and the fed-through input data.

Note: The 8-channel group sent to the analog outputs is defined by the currently chosen ID.

10.11 Gain (GA)

Default: Au

Available settings: Au, 42, IF

Changing the gain can be performed by the microphone (in case the microphone supports this feature) or in the DMC-842. Per default (Au) the DMC-842 reacts on the information received from the microphone. When this feature is supported the DMC-842 switches off its gain control. The choice can also be done manually. Selecting the setting 42 (AES42) the gain stage of the microphone is used, provided it has one. With the setting IF (Interface, means DMC-842) the gain stage of the DMC-842 is used exclusively. In this case a gain of 0 dB is sent as AES42 control data to the microphone.

10.12 Control Pulse (Cp)

Default: On

Available settings: On, Off

The DMC-842 uses so called Control Pulses to send control commands to the microphone. These can be switched on and off per channel by the SELECT keys. With activated Control Pulses (On) the PAR LED is lit.

Note: When DPP is switched off the Control Pulses are automatically deactivated.

10.13 Sync Pulse (SP)

Default: On

Available settings: On, Off

The control of the sample frequency in Mode 2 is done by the use of so called Sync Pulses. These can be switched on and off per channel by the SELECT keys. With activated Sync Pulses (On) the PAR LED is lit.

Note: When DPP is switched off the Sync Pulses are automatically deactivated.

10.14 Special (SPECIAL)

Gives access to 5 special functions which are steered by the SELECT keys 1 to 5. The current state is signaled by the PAR LED. Here the function is directly switched off/on by the according SELECT key, not via key 8.

SELECT 1: **AES Sync for AES input 1** as sync source instead of the AES sync input at the D-sub connector. Default: Off.

SELECT 2: **Force sample rate** for Mode2 microphones. This function can be switched off for microphones that work according to an old AES42 standard. Default: On.

SELECT 3: **Auto SRC**. When this function is active, the SRC in an AES42 input is activated as soon as a signal is applied, which is not sent by a Mode2 microphone. With auto SRC active, the SRC cannot be switched manually. Default: On.

SELECT 4: **Auto CP/SP**. When this function is active, the transmission of control and sync pulses depends on the microphone's properties. In this case, CP and SP cannot be switched manually. Default: On.

SELECT 5: **Display auto dark**. When display auto dark is active, the display goes dark after a couple of seconds without an entry on the front panel. Only the decimal points are still lit. In this state, all buttons are locked and will only be unlocked by keeping a button pressed for a few seconds. Default: Off

11. The Parameter Menu

11.1 General

After pressing the **PAR** button below the rotary encoder the DMC-842 enters the Parameter menu. All eight PAR LEDs light up. Turning the encoder knob scrolls through the available options. After selecting the desired option press and hold the Select key of the channel where the parameters shall be changed. Turning the encoder knob increases and decreases the current parameter value.

All changes are stored automatically. With a second push on the PAR button the Parameter menu is exit.

The Parameter menu grants direct access to settings which are usually handled comfortably by the use of the included Windows software. Additionally it allows to change settings which might become of importance in future revisions of the AES42 standard (including upcoming microphones supporting these settings). But such changes would also be part of an update to the Windows software. In short, the Parameter menu is not intended to be used daily, but as an emergency function, giving access to all settings even without the use of a computer.

The Parameter menu includes a direct display only for the most important functions. The others are configured by a combination of command byte and parameter byte, based on the table found in the latest revision of the AES42 standard, and further information from the corresponding microphone manufacturer. The DMC-842 and the Windows software provide functionality based on the AES42 standard revision of 2006.

Note: Microphones often do not support all the features defined in the standard. Settings performed in the Parameter menu may not be carried out by the microphone, if the specific feature is not supported. The supported features are shown in the Windows software.

11.2 Low Cut (Lo-Cut)

Default: Off

Available settings: Off, 40, 80, 160 (display 0, 1, 2, 3)

This option activates a high-pass (bass removal) within the microphone. This filter can remove rumble, subsonic-noise and other low frequency noises. The cut-off frequency can be set to 40 Hz, 80 Hz and 160 Hz.

11.3 Directivity Pattern (PAttErn)

Default: 0

Available settings: 0 up to 15 (display 0 up to F, hex coded)

The remote-controlled setting of the directivity patterns is one of the highlights of digital microphones. The pattern can be adjusted in 15 steps from omnidirectional (1) to cardioid (7) up to figure of eight (15).

11.4 Pre-Attenuation (AttEnuAtlon)

Default: Off

Available settings: 0 dB, -6 dB, -12 dB, -18 dB (display 0, 1, 2, 3)

To prevent the microphone from overload by extremely loud sound sources an adjustable attenuator is used which can be set up by this option.

11.5 Mute (mutE)

Default: Off

Available settings: On, Off (display 0, 1)

This option allows to mute the microphone output via remote control. Such a function can be useful to prevent gain changes where audio should stop but the gain should stay unchanged, or to dim the level meters of unused channels.

11.5 Peak Limiter (LlmtEr)

Default: Off

Available settings: On, Off (display 0, 1)

A peak limiter removes peaks and short overloads, but can also reduce the dynamic range intentionally.

11.7 Command Type (ComtyPE)

The DMC-842 offers access to the command bytes 1, 2 and 4. The parameter of the command byte can be set like done in the other functions, using Select keys and rotary encoder. The parameter values are displayed hex-coded (00 – FF).

12. Remote Control

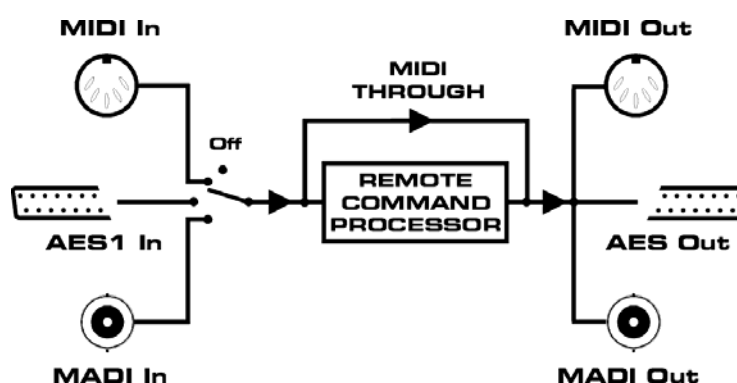
12.1 MIDI

The DMC-842 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 DMC-842 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 21.

The button REMOTE is used to select the source of the MIDI remote control commands: AES, MIDI, MADI, RS232 and Off. The latter is a safety function to prevent any setup-change by any MIDI signal. RS232 is currently not operational.

With an installed I64 MADI Card the DMC-842 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 12.2, 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 DMC-842s. The same is true for remote control via MADI, where MIDI is passed through from device to device automatically via the serial MADI cabling.



12.2 MIDI over MADI

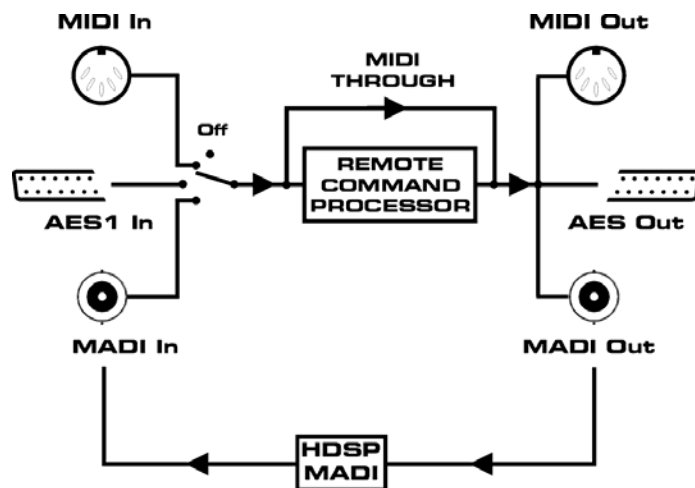
MADI allows for a transmission of 64 audio channels over long distances with a single line – perfect. But what about MIDI? Be it remote control commands or sequencer data, in practice only a single MADI line will not suffice. Therefore RME developed the *MIDI over MADI* technology. The data at the MIDI input are being included into the MADI signal invisibly, and can be collected at the MIDI output of another DMC-842, ADI-8 QS, ADI-6432 / 642 / 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 use 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 DMC-842 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 DMC-842.



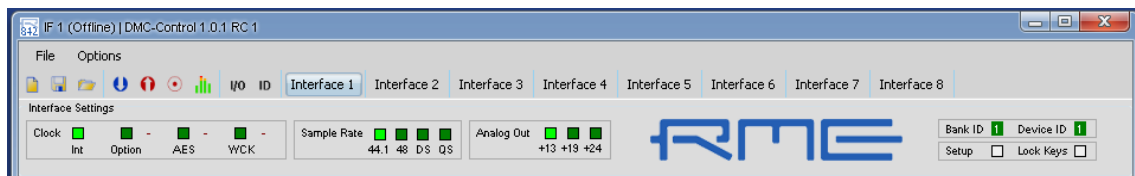
Additional to the remote commands further MIDI data can be transmitted, available then at the DIN MIDI output jack of the DMC-842. However, MIDI signals at the DIN input jack can not travel back to the computer. With input setting MIDI the other direction is active. MIDI data travel from the MADI out to the computer, but not back from the computer via MADI to the DMC-842.

12.3 Remote Control Software

The included Windows software DMC-Control can use any existing MIDI port within the system to perform remote control and status requests of any number of DMC-842s via a simple mouse click. The latest version of the program can be downloaded for free from the RME website. Most appealing is the combination with a HDSP(e) MADI (PCI card), offering a direct control of the DMC-842 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 (see above).

Software download: http://www.rme-audio.de/download/dmc_control.zip

Brief description of the Windows Software DMC-Control



The program requires Microsoft .NET Framework version 2.0 or higher to be installed on the computer. Is this not the case it will be installed automatically.

The program is self-explanatory and very easy to use. It manages up to 8 DMC-842 simultaneously. Names can be assigned to all channels and all devices. DMC-Control not only offers a remote-access to all front panel settings, but also a comfortable configuration of the Setup menu, and direct access to the AES42 features of the connected microphones.

After the start first select a MIDI input and output which is connected to the DMC-842, via **Options - MIDI I/O Setup**.

Using the **Online** button the communication with the DMC-842 is started. The bottom line of the program window shows the current state, like MIDI ports, Online / No Response / Offline etc.

The button **Send Data** allows an offline configuration of the DMC-842 by a single one-time transmission of all settings.

The button **Receive Data** allows a one-time request of all settings of all DMC-842.

Via **File / Save** complete setups with up to 8 DMC-842 can be saved and recalled anytime.

Via MIDI remote control, all front panel controls of the DMC-842 can be locked (**Lock Keys**). An exception is the REMOTE key. In the Off-state the function Lock Keys is deactivated. Therefore a locking of all the controls can be revoked directly at the unit at any time.

The eight identical channels consist of the following elements:

Level Meter

Lock, Sync, Limiter Active and Mode 2 display

Activation of DPP, SRC, Stereo (odd channels only) and M/S

The channel name can be edited

Gain setting

Setting the Directivity Pattern

Choice of Mute and Phase Inversion

Selection of the Pre-Attenuation (PAD), 0, -6, -12 or -18 dB

Choice of the Low Cut cut-off frequency, 0 Hz (Off), 40 Hz, 80 Hz and 160 Hz

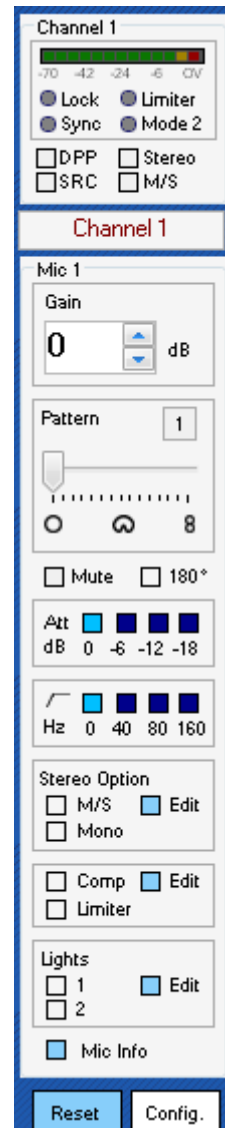
Several options which are available only when using stereo microphones: M/S, Mono, Edit (opens a dialog for the M/S and L/R configuration)

Activation of Limiter and Compressor (microphone based features). Edit opens a dialog to configure several parameters.

Choice of indication light at the microphone: 1 and 2. Edit opens a dialog to set the brightness.

Mic Info: Opens a dialog presenting all the features reported by the microphone as supported.

Reset sets all the settings back to a default state. Config allows for an offline configuration of currently not active (greyed out) parameters.



12.4 RS232

The DMC-842 is already equipped with a serial interface port. Please note that this port is without functionality at this time and reserved for future upgrades.

User's Guide



DMC-842

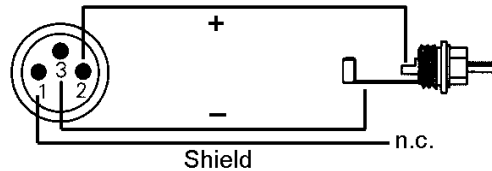
▶ Inputs and Outputs

13. Digital Inputs

13.1 XLR

At the rear side of the DMC-842 there are eight XLR sockets for the AES42 inputs. All inputs are transformer-balanced, ground-free and compatible to AES3. Channel status and copy protection are being ignored.

Thanks to a highly sensitive input stage, SPDIF signals can also be processed by using a simple cable adapter phono/XLR. To achieve this, pins 2 and 3 of a male XLR plug are connected individually to the two pins of a phono plug. The cable shielding is only connected to pin 1 of the XLR - not to the phono plug.




The inputs can be used in any combination, e. g. it is sufficient to connect an input signal only to input 3.

Note: Please note that when using AES input signals the DMC-842 is not able to synchronize on these. It is necessary to either use another Sync reference (D-sub AES1 or word clock), or to activate the sample rate conversion.

The DMC-842 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 ADI-192 DD, an 8-channel universal sample rate and format converter, is highly recommended.

Emphasis

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

 *An Emphasis indication gets lost! This information is neither passed on to the MADI output, nor to any of the AES outputs, and has also no effect on the analog outputs!*

The pin assignment follows international standards. With XLR, pin 2 is + or hot, pin 3 is – or cold, pin 1 is ground. Pin 1 is connected to the chassis directly at the socket (AES48).

13.2 D-Sub - AES/EBU Sync

The input AES 1 (channel 1/2) found on the D-sub connector can not be used for audio with the DMC-842, but as clock source and for remote control. The input is transformer-balanced and ground-free. Thanks to a highly sensitive input stage, a SPDIF signal can also be fed by using a simple cable adapter phono/XLR (see above).

Pinout of the D-sub connector, Inputs

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

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

14. Digital Outputs

14.1 AES/EBU

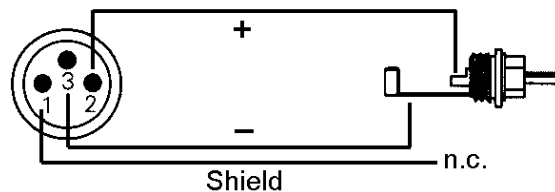
The four AES/EBU outputs are provided on the rear of the DMC-842 via a 25 pin D-sub connector with Tascam pinout (also used by Digidesign). A digital breakout cable will provide 4 male (and 4 female) XLR connectors. Every output is transformer-balanced, ground-free and compatible to all devices with AES/EBU ports.


In normal operation the AES outputs carry the converted analog input signal. When using the I64 MADI Card and activating the option *oP* in the Setup menu *do*, the MADI input data is present at the output instead, see chapter 11.11.

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 DMC-842 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: DMC8

Connecting devices with coaxial SPDIF ports to the DMC-842's outputs 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.



 *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.*

The DMC-842 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.

Pinout of the D-sub connector, Outputs

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

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

14.2 ADAT Optical

The DMC-842 provides two digital outputs in ADAT optical format. In normal operation these ports carry the converted analog input signal. When using the I64 MADI Card and activating the option *oP* in the Setup menu *do*, the MADI input data is present at the output instead, see chapter 11.11. In Single Speed mode both outputs carry the same audio data. With this it is possible to distribute the output signal to two devices (2 x ADAT splitter).

As the ADAT optical signal is physically specified up to 48 kHz only, the DMC-842 automatically activates Sample Split mode (S/MUX) at 88.2 and 96 kHz, distributing the data of one channel to two output channels. The internal frequency stays at 44.1/48 kHz. Therefore the sample clock at the ADAT outputs is only half the frequency of the AES outputs. As interesting as this is – you don't need to think about it. 96 kHz capable ADAT hardware, like all current RME digital interfaces, re-combine the data automatically. The user (and the DAW software) does not see any split data, but just single channels at the expected double sample rate.

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.

The ADAT optical outputs of the DMC-842 are fully compatible to all ADAT optical inputs. A usual TOSLINK cable is sufficient for connection.

ADAT Main

Interface for the first or only device receiving an ADAT signal from the DMC-842. Carries the channels 1 to 8. When sending a Double Speed signal, this port carries the channels 1 to 4. In Quad Speed mode this port carries channels 1 and 2.

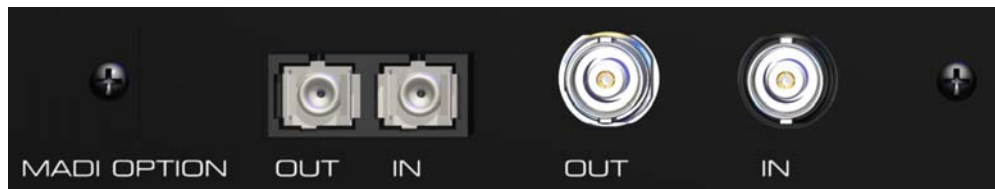
ADAT AUX

In Single Speed mode carries a copy of the data at the Main output. When sending a Double Speed signal, this port carries the channels 5 to 8. In Quad Speed mode this port carries the channels 3 and 4.

14.3 I64 MADI Card

The I64 MADI Card provides the DMC-842 with a 64-channel MADI input and output. Coaxial and optical output operate simultaneously and deliver the same data. The ID determines which MADI channels the DMC-842 will use (see chapter 10.2, 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 features an optical as well as a coaxial MADI input. The input is switched automatically, according to where a valid input signal is detected. Full redundancy is ensured by the automatic input switching, immediately changing to the other input in case of loss of the input signal.



The MADI input will operate as an optional clock source (Clock section, OPTN) as well as a through input. Since each DMC-842 uses only 8 channels, up to 56 channels can be passed through.

This technique is used to serially cascade several DMC-842s. Incoming MADI data is passed through unchanged, only one block of eight channels is replaced, according to the DMC-842's 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.

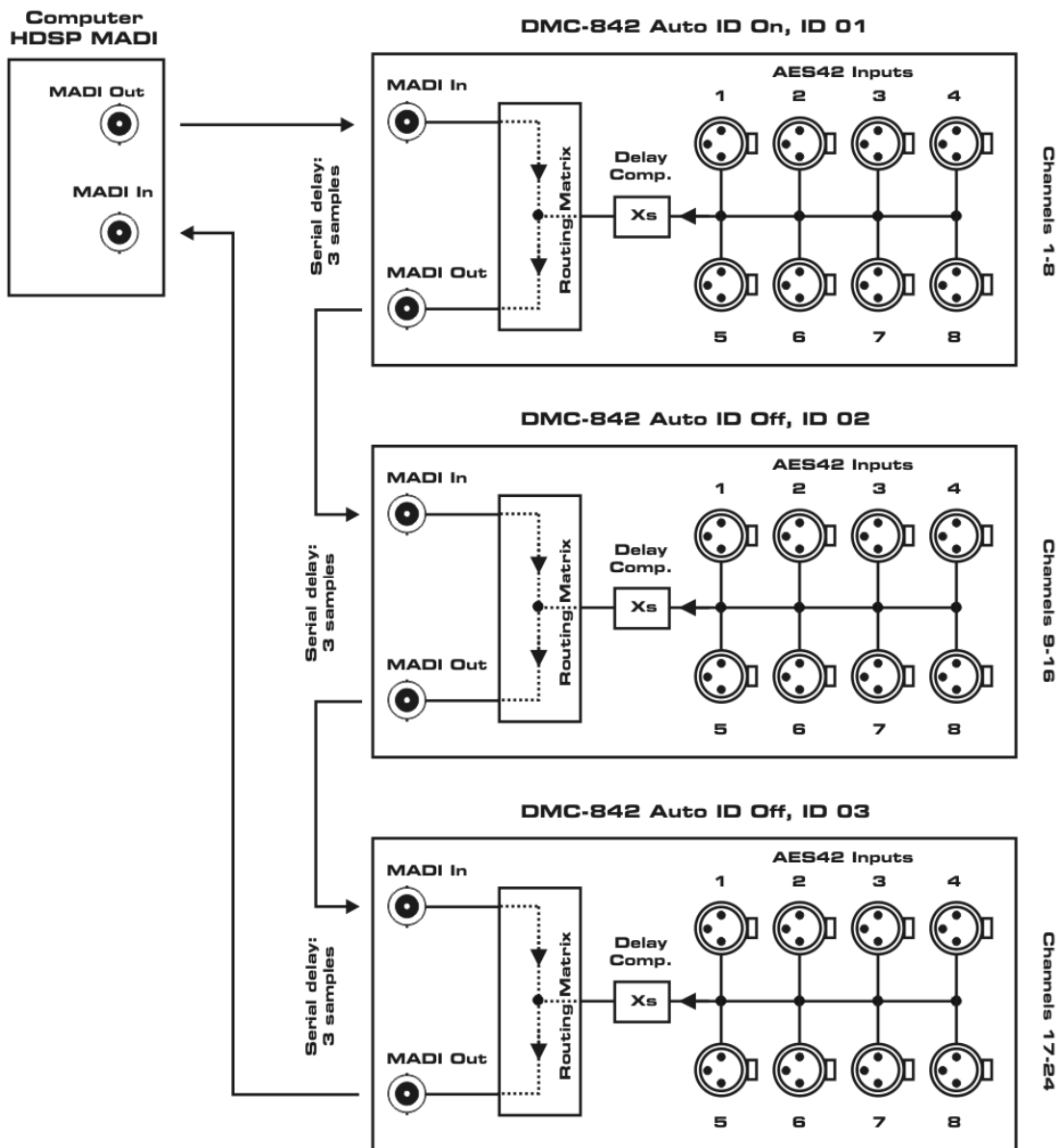
With an installed I64 MADI Card the DMC-842 can be remote controlled via MADI. At the same time MIDI data are transmitted via MADI, see chapter 12.2.

When multiple devices are connected serially, the MADI I/O of each DMC-842 causes a delay of 3 samples. Therefore at the MADI output of the last device, the data of all upstreamed 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 11.3. 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 DMC-842s and activated Delay Compensation.



Delay Compensation has to be manually activated in each unit!



14.4 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 DMC-842s require to activate Delay Compensation manually in each unit.

15. Analog Outputs

The DMC-842 has 8 balanced XLR outputs on the back panel.



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

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

Reference	0 dBFS @	Analog Headroom
+24	+24 dBu	3 dB
+19	+19 dBu	8 dB
+13	+13 dBu	14 dB

The DMC-842 delivers an undistorted maximum level of +27 dBu. The referencing is based on full scale of the DA-converters. 0 dBFS will be reached at +13, +19 or +24 dBu output level respectively.

The different output levels guarantee optimal conversion results, while still being compatible to any attached analog equipment. Note that ANALOG OUT has no influence on the technical specifications – distortion, frequency response and signal to noise ratio do not change, or rather exactly like the analog overall gain.

Set to +13 dBu and +19 dBu the DMC-842 is fully compatible to all RME devices having +4 dBu and Lo Gain as input reference.

Reference	0 dBFS @	Headroom @ +4 dBu
Lo Gain	+19 dBu	15 dB
+4 dBu	+13 dBu	9 dB

Set to +24 dBu the DMC-842 is compatible to SMPTE (+24 dBu @ 0 dBFS, +4 dBu with 20 dB of headroom).

16. Word Clock

16.1 Word Clock Input and Output

SteadyClock 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 BNC output (see section 18.10).

Input

The DMC-842's 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 DMC-842 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 (0.3V) input level is 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 DMC-842's 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 DMC-842 can be distributed to other devices by using the word clock output. With this the usual T-adaptor can be avoided, and the DMC-842 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 DMC-842 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.

16.2 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 DMC-842 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 reality was different, the Superclock proved to be much more critical than word clock. A square wave signal of 11 MHz distributed to several devices - this simply means to fight with high frequency technology. Reflections, cable quality, capacitive loads - at 44.1 kHz these factors may be ignored, at 11 MHz they are the end of the clock network. Additionally it was found that a PLL not only generates jitter, but also 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. In the end Superclock did not become a commonly accepted standard.

The actual end of these problems is offered by the **SteadyClock** technology of the DMC-842. 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.

16.3 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 examples, when the 75 Ohms are built into the device and cannot be switched off. In this case the network load is often 2 x 75 Ohms, and the user is forced to buy a special word clock distributor. Note that such a device is generally recommended for larger studios.

The DMC-842's word clock input can be high-impedance or terminated internally, ensuring maximum flexibility. If termination is necessary (e.g. because the DMC-842 is the last device in the chain), push the switch at the back (see chapter 15.1).

In case the DMC-842 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 DMC-842, we recommend to not pass the input signal via T-adapter, but to use the DMC-842's word clock output instead. Thanks to SteadyClock, the input signal will both be freed from jitter and - in case of loss or drop out – be held at the last valid frequency.

17. MIDI

The DMC-842 has a standard MIDI input and output, a 5-pin DIN jack each. The MIDI I/O is used for:

- remote control of the DMC-842, see chapter 12.1
- transmission of MIDI data and remote control commands over MADI, in case the optional I64 MADI Card has been fitted, see chapter 12.2.

User's Guide



DMC-842

▶ **Technical Reference**

18. Technical Specifications

18.1 Analog

DA-conversion

- Resolution: 24 Bit
- Signal to Noise ratio (SNR) @ +24 dBu, 44.1 kHz: 116,0 dB RMS unweighted, 119 dBA
- Signal to Noise ratio (SNR) @ +19 dBu: 116 dB RMS unweighted, 119 dBA
- Signal to Noise ratio (SNR) @ +13 dBu: 116 dB RMS unweighted, 119 dBA
- Frequency response @ 44.1 kHz, -0.5 dB: 5 Hz – 22 kHz
- Frequency response @ 96 kHz, -0.5 dB: 5 Hz – 34 kHz
- Frequency response @ 192 kHz, -1 dB: 5 Hz - 50 kHz
- THD: < -104 dB, < 0.00063 %
- THD+N: < -100 dB, < 0.001 %
- Channel separation: > 110 dB

18.2 Digital Inputs

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
- 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)

AES1, D-sub

- 1 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)

AES42, XLR

- 8 x XLR, transformer-balanced, galvanically isolated
- Compatible to AES3 und AES/EBU
- Technical specifications like AES1, plus
- Digital Phantom Power (10 V), switchable per input
- Transmission of Sync and Control Pulses

18.3 Digital Outputs

AES/EBU

- 4 x, transformer-balanced, galvanically isolated, according to AES3-1992
- Output voltage 4.5 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 Ohms: 4.0 Vpp
- Impedance: 10 Ohms
- Frequency range: 27 kHz – 200 kHz

I64 MADI Card

- Coaxial via BNC, 75 Ohms, 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

18.4 Digital

- Clocks: Internal, AES 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 DA-conversion: near zero
- PLL ensures zero dropout, even at more than 100 ns jitter
- Supported sample rates: 28 kHz up to 200 kHz

18.5 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)

18.6 General

- Power supply: Internal switching PSU, 100 - 240 V AC, 60 Watts
- Typical power consumption: 24 Watts
- Maximum power consumption: < 40 Watts
- Dimensions including rack ears (WxHxD): 483 x 88 x 242 mm (19" x 3.46" x 9.5")
- Dimensions without rack ears/handles (WxHxD): 436 x 88 x 235 mm (17.2" x 3.46" 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

18.7 Firmware

The DMC-842 is internally based on programmable logic. By re-programming of a little component called Flash-PROM, both function and behavior of the unit can be changed at any time.

At the time of writing this manual the unit is shipped with firmware version 2.10. The firmware version is displayed after power on for about one second on the Gain displays of the DMC-842. The current version of firmware and remote software support the AES42 standard revision of 2006.

The I64 MADI Card has its own firmware. I64 MADI Cards with the latest firmware have a sticker with the firmware version date code on them (20070217). The firmware of the I64 MADI Card can be updated in the factory. Please contact the RME support or your local dealer.

18.8 MADI User Bit Chart

- RS-232: channels 1 to 9
- ADC: channel 19
- MIDI: channel 56 (48k) / 28 (96k)

18.9 Connector Pinouts

The 25 pin 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.

XLR sockets

The XLR connectors are wired according to international standards:

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

Note on the analog outputs: The output circuitry does not operate in a servo-balanced way. When connecting unbalanced devices make sure that pin 3 of the XLR output stays unconnected.

RS-232

The 9-pin D-Sub Connector is wired according to industry standard:

Signal	In (Rx)	Out (Tx)	GND	NC
Sub-D	2	3	5	9

Internally connected are pins 1 with 6 and 4, and pin 7 with 8.

Note that currently RS-232 is not functional.

19. Technical Background

19.1 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-existing, 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 allows to transmit 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.

19.2 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 DMC-842, 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 DMC-842 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 DMC-842'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 DMC-842 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 DMC-842 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.

19.3 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 DMC-842 uses a high-class DA-converter from Texas Instruments, offering exceptional Signal to Noise and distortion figures. Its digital filter delays the analog output signals by 29 samples. The exact delays caused by the DA-conversion of the DMC-842 are:

Sample frequency kHz	44.1	48	88.2	96	176.4	192
DA (29 x 1/fs) ms	0,66	0,60	0,33	0,30	0,16	0,15

These values are about a quarter of those available from former DA-converters. They represent an important step in further reducing the latency in the computer-based recording studio. At Quad Speed the added latency can simply be ignored. And with the ADI-8 QS, a matching AD-converter with only 5 up to 10 samples delay is ready to turn 'analog digital monitoring' into real analog-style monitoring.

19.4 DS - Double Speed

When activating the *Double Speed* mode the DMC-842 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 DMC-842 automatically uses *Sample Multiplexing* in DS mode. One channel's data is distributed to two channels according to the following table:

Original	1	2	3	4	5	6	7	8
DS Signal	1/2	3/4	5/6	7/8	1/2	3/4	5/6	7/8
Port	1	1	1	1	2	2	2	2

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.

19.5 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 DMC-842 is limited to 4 channels at the ADAT outputs.

The AES outputs provide 192 kHz as Single Wire only.

19.6 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).

19.7 MADl Basics

MADl, the serial **M**ultichannel **A**udio **D**igital Interface, 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 fiber 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 labeled, these are always (!) glass fiber cables. Plastic fiber cables (POF, plastic optical fiber) 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 fiber (8 μm). By the way, due to the wave-length of the light being used (1300 nm) the optical signal is invisible to the human eye.

19.8 SteadyClock

The SteadyClock technology of the DMC-842 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.

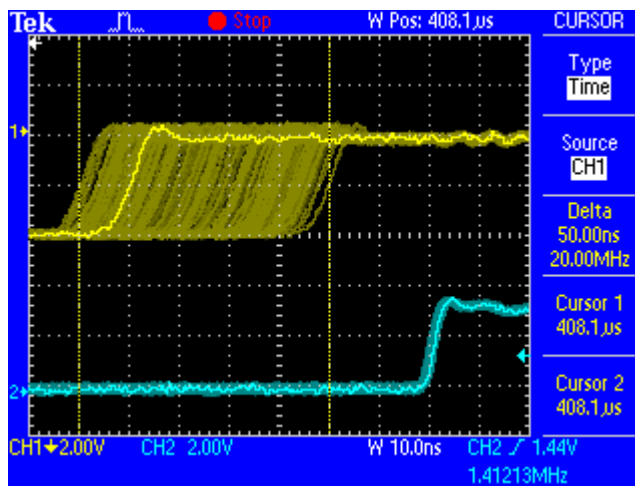
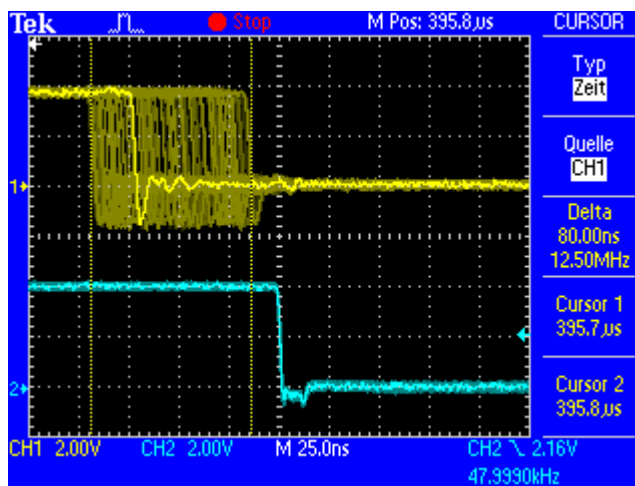
Usually a clock section consists of an analog PLL for external synchronization and several quartz oscillators for internal synchronization. SteadyClock requires only one quartz, using a frequency not equalling digital audio. Latest circuit designs like hi-speed digital synthesizer, digital PLL, 100 MHz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

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

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

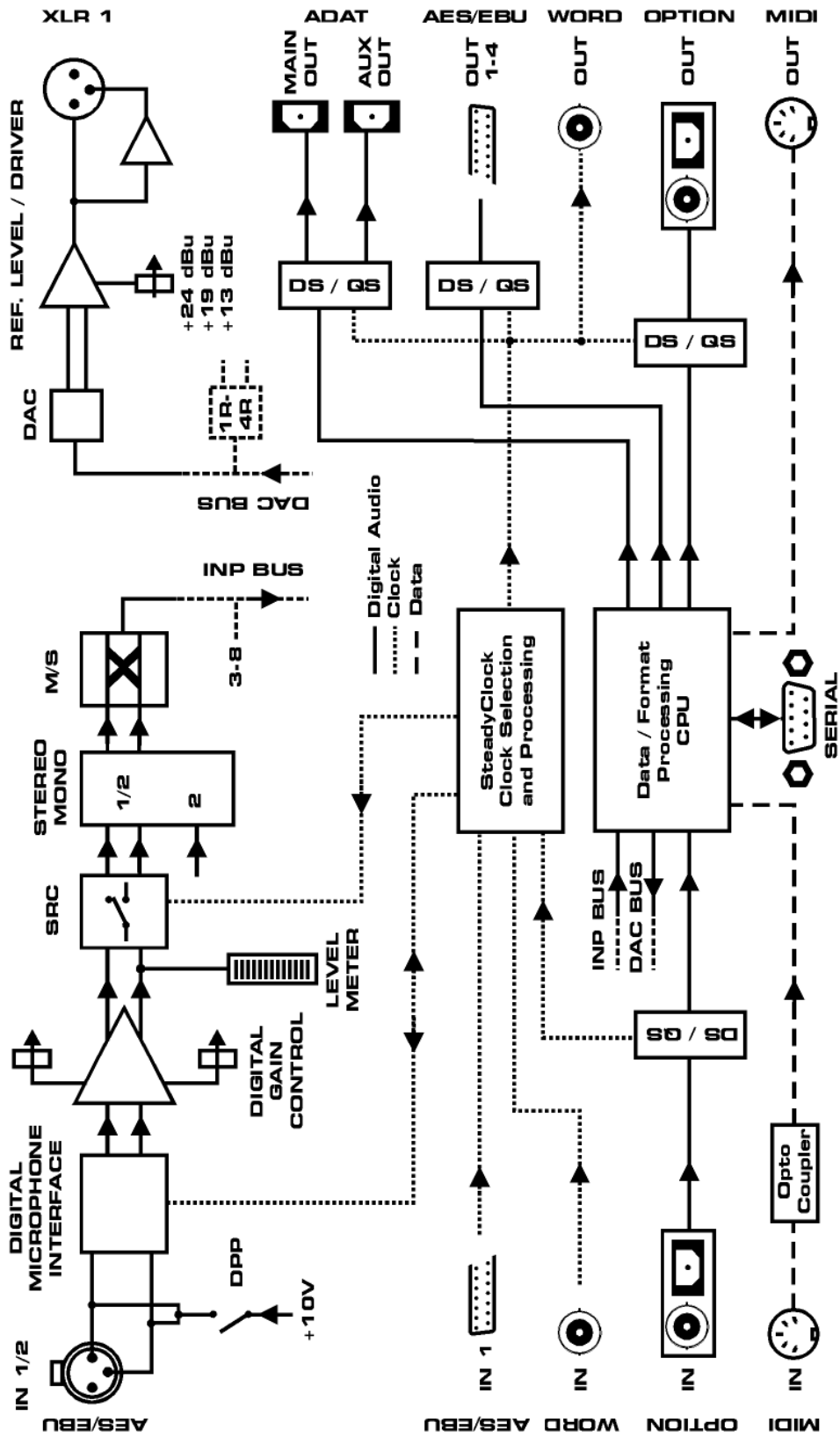
Using the other input sources of the DMC-842, word clock and AES/EBU, you'll most probably never experience such high jitter values. But SteadyClock is not only ready for them, it would handle them just on the fly.

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 DMC-842's word clock output. It is also used to clock the digital outputs MADI and AES/EBU.

20. Block Diagram



21. MIDI Implementation DMC-842

21.1 Basic SysEx Format

<u>Value</u>	<u>Name</u>
F0h	SysEx header
00h 20h 0Dh	MIDITEMP manufacturer ID
42h	Model ID (DMC-842)
00h..77h, 7Fh	Bank number / device ID (7Fh = broadcast, all IDs)
mm	Message type
nn	Parameter number (see table 1)
oo	Databyte
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.

21.2 Message Types

<u>Value</u>	<u>Name</u>
10h	Request interface value
11h	Request level meter data
12h	Request microphone data (replaced by separate values for each channel)
40h	Request microphone data input 1
41h	Request microphone data input 2
42h	Request microphone data input 3
43h	Request microphone data input 4
44h	Request microphone data input 5
45h	Request microphone data input 6
46h	Request microphone data input 7
47h	Request microphone data input 8
20h	Set interface value
22h	Set microphone value
30h	Interface value response
31h	Level meter data response
32h	Microphone data response

Request Interface Value

Format: F0 00 20 0D 42 (bank no. / dev ID) 10 F7

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

Interface 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.

Set Interface Value

Sets any number of parameters.

nn / oo can be repeated freely.

Request Microphone Data

Format: F0 00 20 0D 42 (bank no. / dev ID) (40..47) F7

This string triggers a complete dump of AES42 data of the microphone at input 1 to 8.

Microphone Data Response

After being triggered by receiving a request microphone data command, device sends a string of one channel's complete AES42 microphone data. Message type is set to 32h.

Set Microphone Value

Sets any number of parameters.

AES42 commands are transmitted in groups of 4 Bytes (nn1, nn2, oo1, oo2).

Request Level Meter Data

Format: F0 00 20 0D 42 (Bank no. / dev ID) 11 F7

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

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 42 (bank no. / dev ID) 31 (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.

21.3 Table

No.	No.	Name	Set Val.	Val. Resp.	Databytes
00h	0	Ch. 1 Gain	x	x	0..63 = 0..+63dB, 1dB steps
01h	1	Ch. 1 settings	x	x	(see channel settings table)
02h	2	Ch. 2 Gain	x	x	0..63 = 0..+63dB, 1dB steps
03h	3	Ch. 2 settings	x	x	(see channel settings table)
04h	4	Ch. 3 Gain	x	x	0..63 = 0..+63dB, 1dB steps
05h	5	Ch. 3 settings	x	x	(see channel settings table)
06h	6	Ch. 4 Gain	x	x	0..63 = 0..+63dB, 1dB steps
07h	7	Ch. 4 settings	x	x	(see channel settings table)
08h	8	Ch. 5 Gain	x	x	0..63 = 0..+63dB, 1dB steps
09h	9	Ch. 5 settings	x	x	(see channel settings table)
0Ah	10	Ch. 6 Gain	x	x	0..63 = 0..+63dB, 1dB steps
0Bh	11	Ch. 6 settings	x	x	(see channel settings table)
0Ch	12	Ch. 7 Gain	x	x	0..63 = 0..+63dB, 1dB steps
0Dh	13	Ch. 7 settings	x	x	(see channel settings table)
0Eh	14	Ch. 8 Gain	x	x	0..63 = 0..+63dB, 1dB steps
0Fh	15	Ch. 8 settings	x	x	(see channel settings table)
10h	16	Setup 1	x	x	(see setup 1 table)
11h	17	Setup 2	x	x	(see setup 2 table)
12h	18	Channel Lock / Sync 1		x	(see channel lock / sync table)
13h	19	Channel Lock / Sync 2		x	(see channel lock / sync table)
14h	20	Channel Lock / Sync 3		x	(see channel lock / sync table)
15h	21	Channel Lock / Sync 4		x	(see channel lock / sync table)
16h	22	Host Lock / Sync		x	(see lock / sync table)
17h	23	Setup 3		x	(see setup 3 table)
18h	24			x	(reserved)
19h	25			x	(reserved)
1Ah	26			x	(reserved)
1Bh	27			x	(reserved)
1Ch	28			x	(reserved)

		Channel Settings	MSB / 7	
			6	0
			5	(reserved)
			4	Sync Pulse: 0 = off, 1 = on
			3	Control Pulse: 0 = off, 1 = on
			2	SRC: 0 = off, 1 = on
			1	M/S: 0 = off, 1 = on (set only ch. 1, 3, 5, 7)
			0	Stereo: 0 = off / 2 x mono, 1 = stereo
			LSB / 0	DPP: 0 = off, 1 = on

10h		Setup 1	MSB / 7		0
			6	MSB / 1	Analog Output Level: 0 = +13dBu, 1 = +19dBu,
			5	LSB / 0	Analog Output Level: 2 = +24dBu
			4	MSB / 1	Clock Select: 0 = int., 1 = Option,
			3	LSB / 0	Clock Select: 2 = AES, 3 = WCK
			2	MSB / 1	Clock Range: 0 = single speed, 1 = ds, 2 = qs
			1	LSB / 0	Clock Range
		(d.c. for clock sel > 0)	LSB / 0		Int. Freq.: 0 = 44.1kHz, 1 = 48kHz

11h		Setup 2	MSB / 7		0
			6		Auto-Device: 0 = off, 1 = on
			5		Delay Compensation: 0 = off, 1 = on
			4		Digital Output Signal: 0 = i64, 1 = mic
			3		Clock Out: 0 = always single speed, 1 = Fs
			2		Follow Clock: 0 = off, 1 = on
			1		Peak Hold: 0 = off, 1 = on
			LSB / 0		Lock Keys: 0 = unlock, 1 = lock

12h		Channel Lock / Sync 1	MSB / 7		0
			6		(reserved)
		(Request only)	5		Input 2 RX AES42: 0 = Mode 1, 1 = Mode 2
		(Request only)	4		Input 1 RX AES42: 0 = Mode 1, 1 = Mode 2
		(Request only)	3		Input 2 Sync: 0 = no sync, 1 = sync
		(Request only)	2		Input 2 Lock: 0 = unlock, 1 = lock
		(Request only)	1		Input 1 Sync: 0 = no sync, 1 = sync
		(Request only)	LSB / 0		Input 1 Lock: 0 = unlock, 1 = lock

13h		Channel Lock / Sync 2	MSB / 7		0
			6		(reserved)
		(Request only)	5		Input 4 RX AES42: 0 = Mode 1, 1 = Mode 2
		(Request only)	4		Input 3 RX AES42: 0 = Mode 1, 1 = Mode 2
		(Request only)	3		Input 4 Sync: 0 = no sync, 1 = sync
		(Request only)	2		Input 4 Lock: 0 = unlock, 1 = lock
		(Request only)	1		Input 3 Sync: 0 = no sync, 1 = sync
		(Request only)	LSB / 0		Input 3 Lock: 0 = unlock, 1 = lock

14h		Channel Lock / Sync 3	MSB / 7		0
			6		(reserved)
		(Request only)	5		Input 6 RX AES42: 0 = Mode 1, 1 = Mode 2
		(Request only)	4		Input 5 RX AES42: 0 = Mode 1, 1 = Mode 2
		(Request only)	3		Input 6 Sync: 0 = no sync, 1 = sync
		(Request only)	2		Input 6 Lock: 0 = unlock, 1 = lock
		(Request only)	1		Input 5 Sync: 0 = no sync, 1 = sync
		(Request only)	LSB / 0		Input 5 Lock: 0 = unlock, 1 = lock

15h		Channel Lock / Sync 4	MSB / 7		0
			6		(reserved)
		(Request only)	5		Input 8 RX AES42: 0 = Mode 1, 1 = Mode 2
		(Request only)	4		Input 7 RX AES42: 0 = Mode 1, 1 = Mode 2
		(Request only)	3		Input 8 Sync: 0 = no sync, 1 = sync
		(Request only)	2		Input 8 Lock: 0 = unlock, 1 = lock
		(Request only)	1		Input 7 Sync: 0 = no sync, 1 = sync
		(Request only)	LSB / 0		Input 7 Lock: 0 = unlock, 1 = lock

16h		Host Lock / Sync	MSB / 7		0
			6		(reserved)
		(Request only)	5		WCK Sync: 0 = no sync, 1 = sync
		(Request only)	4		WCK Lock: 0 = unlock, 1 = lock
		(Request only)	3		AES Sync: 0 = no sync, 1 = sync
		(Request only)	2		AES 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

17h		Setup 3	MSB / 7		0
			6		(reserved)
			5		(reserved)
			4		(reserved)
			3		(reserved)
			2	MSB / 1	Gain Target: 0 = auto, 1 = AES42 only,
			1	LSB / 0	Gain Target: 2 = DMC only
			LSB / 0		Analog Output Signal: 0 = i64, 1 = mic

Request Level Meter Data

		Levelmeter Data	MSB / 7		0
			6		RX AES42: 0 = limiter not active, 1 = lim. Act.
			5		Input Sync: 0 = no sync, 1 = sync
			4		Input Lock: 0 = unlock, 1 = lock
			3	MSB / 3	Level: 0 = < -70dBFS peak
			2	/ 2	Level: 1..12 = < -60 / -50 / -42 / -36 / -30 / -24 /
			1	/ 1	Level: -18 / -12 / -6 / -3 / -1 / -0.1 dBFS
			LSB / 0	LSB / 0	Level: 13 = > -0.1 dBFS (over)