

User's Guide



OctaMic II

SteadyClock™

Professional Mic / Line Preamp and AD-Converter
8-Channel Microphone / Line Preamp with Line Outputs
8-Channel Analog to AES / ADAT Interface
24 Bit / 192 kHz Digital Audio

 **ADAT®**
OPTICAL AES-3
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



OctaMic II

▶ General

1. Introduction

The OctaMic II's innovative concept allows for amplification and digitization of ALL analog signal sources. Be it high-level stage signals, typical studio signals, lower level and high-impedance instruments, or dynamic, condenser or ribbon microphones: OctaMic II understands them all – in a way that is simply thrilling.

When developing the OctaMic II we used all our experience, and also the experience of our customers, to create a unique, excellent and high-quality unit. As successor of the OctaMic D the device offers the approved quality of its predecessor, adding lots of improvements in detail.

2. Package Contents

Please check that your OctaMic II package contains each of the following:

- OctaMic II
- Power cord
- Manual
- 1 optical cable (TOSLINK), 2 m

3. Brief Description and Characteristics

- 8 balanced XLR/TRS mic/line inputs
- 54 dB gain range
- Analog input level from –40 dBu up to +21 dBu
- Large frequency range (200 kHz) with special EMI input filtering
- Input impedance: XLR 2 kOhm, TRS 5 kOhm
- Signal to noise ratio (SNR): 129 dB EIN @150 Ohm
- THD: < 0.0005 % @ 30 dB Gain
- Channel separation: > 110 dB
- Frequency response –0.5 dB: 5 Hz - 200 kHz
- Line Out: 1/4" TRS (6.3 mm stereo jack), servo-balanced
- Maximum output level: +21 dBu
- Output impedance: 75 Ohm
- Output level switchable Hi Gain / +4 dBu / -10 dBV
- Word clock input
- 4 x AES/EBU Out per D-sub, 8 channels @ 192 kHz
- 2 x ADAT Out, 8 channels @ 96 kHz
- SNR ADC: > 110 dBA
- Sample rate range ADC: 28 kHz – 200 kHz
- THD AD: < 0.00032 %, < -110 dB

4. Accessories

RME offers several optional components for the OctaMic II:

Part Number	Description
OK0050	Optical cable, Toslink, 0.5 m
OK0100	Optical cable, Toslink, 1 m
OK0200	Optical cable, Toslink, 2 m
OK0300	Optical cable, Toslink, 3 m
OK0500	Optical cable, Toslink, 5 m
OK1000	Optical cable, Toslink, 10 m
BO25MXLR4M4F1PRO	Digital Breakout Cable Pro, AES/EBU 25-pin D-sub to 4 x XLR male + 4 x XLR female, 1m
BO25MXLR4M4F3PRO	Digital Breakout Cable Pro, AES/EBU 25-pin D-sub to 4 x XLR male + 4 x XLR female, 3 m
BO25MXLR4M4F6PRO	Digital Breakout Cable Pro, AES/EBU 25-pin D-sub to 4 x XLR male + 4 x XLR female, 6 m
BO25M25M1PRO	Digital D-sub Cable Pro, AES/EBU 25-pin D-sub to 25-pin D-sub, 1m
BO25M25M3PRO	Digital D-sub Cable Pro, AES/EBU 25-pin D-sub to 25-pin D-sub, 3m
BO25M25M6PRO	Digital D-sub Cable Pro, AES/EBU 25-pin D-sub to 25-pin D-sub, 6m

5. Warranty

Each individual OctaMic II 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 OctaMic II. The general terms of business drawn up by Audio AG apply at all times.

6. Appendix

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

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

Distributor:

Audio AG, Am Pfanderling 60, D-85778 Haimhausen, Tel.: (49) 08133 / 91810

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 OctaMic II.

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



OctaMic II

▶ Usage and Operation

7. Front Panel Controls

7.1 Displays

+48V (LED) lights up when phantom power is active.

The **CLIP** LED has been designed to act like the OVR LEDs of the ADI-8 series. It lights up 2 dB before the chosen reference level plus a headroom of 9 dB. At Hi Gain the LED lights up at +17 dBu output level, selecting +4 dBu it lights up at +11 dBu.


SIG (Signal) indicates the presence of an input signal. The LED has a detection range of more than 50 dB using multiple brightness states. With this, SIG acts as useful level control, helping to set GAIN correctly.



7.2 Controls

GAIN allows for a step-less adjustment of the amplification between +6 dB and +60 dB.

+48V (switch) activates phantom power. Phantom power should only be activated when using condenser microphones which require such a power supply, and only on the specific channel.

 *Connecting and disconnecting microphones while phantom power is active causes a high voltage surge, which can destroy the microphone input stage! Switch phantom power off before connecting/disconnecting any external device.*

The OctaMic II turns on the phantom power smoothly during one second, from 0 to 48 Volts. This technique is advantageous for the connected microphone as well as the OctaMic II. The phantom power of the OctaMic II is short-circuit proof. With a maximum load on all eight channels the internal voltage does not drop below 47 Volts.

LO CUT activates a high-pass (bass removal) with 18 dB per octave, at a cut-off-frequency of 80 Hz. This filter can remove rumble, subsonic-noise and other low frequency noises. The LO CUT of the OctaMic II shows a slow roll-off in the frequency response, without any resonance rise, and provides low THD figures. At a low 20 Hz, where the fundamental signal is already attenuated by 34 dB, the distortion is around 0.13 %.

PHASE changes the polarity (180°). Phase cancellations and sound changes can be caused by using multiple microphones at different places, or wrongly soldered cables. In such cases PHASE can eliminate the error by adding an additional phase inversion.

Clip Hold is activated by pressing the key for two seconds. As soon as an overload is detected, the corresponding Clip LED begins to flash once per second. With this, a momentary overload stays visible for a longer time. Pressing the key once resets the Clip display. Pressing the key again for two seconds deactivates the Clip Hold mode.



Hi Gain / +4 dBu / -10 dBV: Defines the reference level of the Line Level Outputs, equalling a full scale level of the AD-converters. See chapter 10.2.

8. Rear Panel

8.1 Connectors

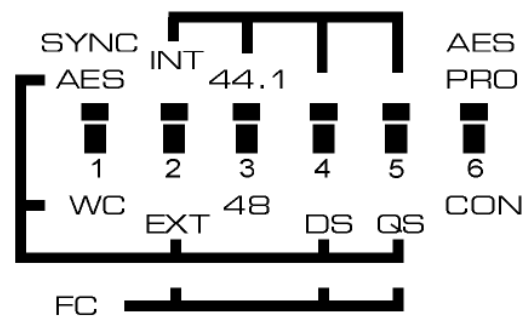
MICROPHONE / LINE INPUTS: 8 8 Neutrik XLR / TRS combo jacks. Thanks to the servo balanced designs and a high maximum input level (maximum input level XLR +14 dBu, TRS +21 dBu) the inputs are universally applicable.

LINE LEVEL OUTPUTS: 8 TRS (stereo) jacks. The electronic output stage is built in a servo balanced design, handling monaural (unbalanced) and stereo jacks (balanced) correctly.

IEC receptacle for power connection. The specially developed, internal hi-performance switch mode power supply lets the OctaMic II 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.

8.2 DIP Switches

The DIP switches are used to configure the OctaMic II's digital section. The following diagram, also printed on the back of the unit, shows the meaning of all possible switch positions.



DIP Switch	Function
1	External synchronization source AES (D-sub) or word clock (BNC)
2	Clock internal (Master) or external (Slave)
3	Internal Clock 44.1 kHz or 48 kHz
4	Activates Double Speed Mode*
5	Activates Quad Speed Mode*
6	AES output state Professional or Consumer

*Note on DIP switch 4/5:

At internal clock, the switches DS and QS multiply the value set with switch 3 by a factor of 2 or 4. So if switch 3 is set to 48 kHz, switch 4 will turn it into 96 kHz, switch 5 turns it into 192 kHz.

At external clock switch 3 is of no meaning, because the unit is synchronized to the incoming clock. However, switch 4 and 5 pre-define the frequency range to Single Speed, Double Speed or Quad Speed. For example if the OctaMic shall operate at 176.4 or 192 kHz, switch 5 has to be set to the lower position. The OctaMic will now generate an output signal in the Quad speed range (176.4 or 192 kHz), even with a word clock input signal of only 44.1 kHz, or an AES input signal of 96 kHz.

With external synchronization active (switch 2 down) and both switches 4 and 5 in lower position, the function *Follow Clock* (FC) is active. In this case the OctaMic II follows the input clock 1:1. A further configuration of the current sample rate range (Single, Double or Quad Speed) is not required.

9. External Synchronization

The digital inputs of the OctaMic II are used for external synchronization only. In case the clock shall not be generated internally (operation mode Master), an external synchronization (operation mode Slave) is available via word clock or AES (SPDIF).

The SteadyClock technology of the OctaMic II guarantees exceptional performance in all clock modes. Thanks to a highly efficient jitter suppression, the AD-conversion always operates on highest sonic level, being completely independent from the quality of the incoming clock signal. When the current word clock source fails, the last valid sample rate will be held automatically.

9.1 Word Clock - BNC

The word clock input is activated by DIP switches 1 and 2. Both switches must be set to their lower position.

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 beside the BNC socket. Use a small pencil or similar and carefully push the blue switch so that it snaps into its lock position. Another push will release it again and de-activate the termination.

Due to the outstanding clock control a synchronization of the output signal to the input signal is not only possible at identical sample rates, but also at half, quarter, double and quad sample rates.

Example 1: DIP switch 3/4/5 in upper position results in a sample rate of 44.1 kHz. The external synchronization source (word clock or AES) can now be 44.1 kHz, 88.2 kHz or 176.4 kHz.

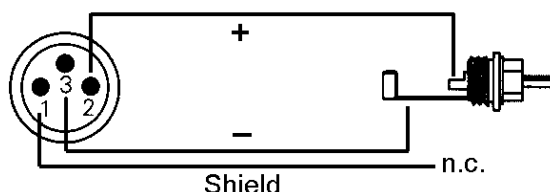
Example 2: DIP switch 3/5 in lower position results in a sample rate of 192 kHz. The external synchronization (word clock or AES) source can now be 48 kHz, 96 kHz or 192 kHz.

9.2 AES – D-Sub

Using the D-sub connector, an AES, AES/EBU or SPDIF signal can be used for synchronization. Set DIP switch 1 to the upper position and DIP switch 2 to the lower position.

The OctaMic II's synchronization input uses AES 1 (see chapter 11.1). 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. 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.



With AES synchronization is possible not only at identical sample rates, but also at half, quarter, double and quad sample rates.

User's Guide



OctaMic II

▶ Inputs and Outputs

10. Analog Inputs / Outputs

10.1 Mic / Line In

The OctaMic has 8 balanced Mic and Line inputs via 1/4" TRS (stereo) and XLR combo jacks on the rear panel. The electronic input stage is built in a servo balanced design which handles unbalanced and balanced signals correctly, automatically adjusting the level reference.

XLR

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).



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

The OctaMic II offers an adjustable amplification from +6 dB up to +60 dB. This equals a sensitivity of +14 dBu down to –40 dBu, referenced to full scale of the AD-converter.

The soft switching, hi-current phantom power (48 Volt) provides a professional handling of condenser microphones. The usage of a hi-end integrated circuit (That 1510) guarantees outstanding sound quality, lowest THD, and maximum Signal to Noise ratio in any gain setting.

The OctaMic II's 'overall' amplification from analog input to analog output depends on the analog output reference setting. The EIN is not affected by this setting, since the S/N ratio scales 1:1 with the output amplification.

The input impedance is 2 kOhm.

TRS

The pin assignment follows international standards. With TRS, tip is + or hot, ring is – or cold.



When using unbalanced cables with stereo TRS jacks, the 'ring' contact of the cable's jack should be connected to pin 1 (ground). Otherwise noise may occur, caused by the unconnected negative input of the balanced input.

The TRS jacks have a fixed level attenuation of 7 dB. Based on the adjustable amplification from +6 dB up to +60 dB, the sensitivity is +21 dBu down to –33 dBu, referenced to full scale of the AD-converter. Therefore the TRS inputs are true full level Line inputs, and the unit can also be used as Line amplifier.

The TRS jacks are free of phantom power. The unbalanced input impedance is 5 kOhm.

10.2 Line Out

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

The pinout follows international standards. Tip + or hot, ring – or cold.

To maintain an optimum level for devices connected to the analog outputs, the OctaMic includes a switch which allows to change the reference level of all 8 outputs simultaneously.

The OctaMic II can generate a maximum level of +21 dBu without distortion. However, the CLIP LED has been designed to act like the OVR LEDs of the ADI-8 series. It lights up 2 dB before the reference level selected on the back of the unit, plus a headroom of 9 dB, is reached. At Hi Gain the LED lights up at +17 dBu output level, selecting +4 dBu it lights up at +11 dBu, selecting –10 dBV it lights up at 0 dBV.

Setting	Reference	Clip LED	True Clip	ADC Level
Hi Gain	+19 dBu	+17 dBu	+21 dBu	-2 dBFS
+4 dBu	+13 dBu	+11 dBu	+15 dBu	-2 dBFS
-10 dBV	+2 dBV	0 dBV	+4 dBV	-2 dBFS

This also means that the CLIP LED lights up 4 dB before the OctaMic II actually reaches the maximum level. Such an additional headroom is considered to be useful in real world operation.

The chosen reference level has no meaning for the digital outputs. The AD-conversion is designed for a level of –2 dBFS as soon as the Clip LED lights up.

Selecting +4 dBu the output signal is attenuated by 6 dB, so for the same output level the amplification has to be increased via GAIN. With this trick the OctaMic reaches the maximum signal to noise ratio on +4 dBu based inputs (like our ADI-8 series), because microphone preamps have better EIN values at higher amplification. In case of an extreme recording situation, where the gain of the OctaMic is no longer sufficient, selecting Hi Gain will again provide the highest amplification possible.

The same is true and even more efficient at –10 dBV. In this case the output level is reduced by around 14 dB – the same happens to the basic noise of the unit!

11. Digital Outputs

11.1 AES/EBU

The four AES/EBU outputs are provided on the rear of the OctaMic II 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.

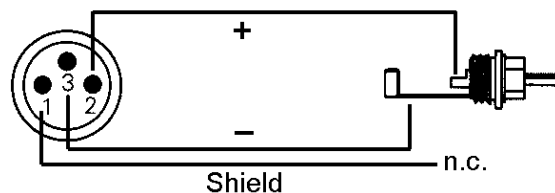
Besides the audio data, digital signals in SPDIF or AES/EBU format contain a channel status coding, which is being used for transmitting further information. The output signal coding of the OctaMic II has been implemented according to AES3-1992 Amendment 4:


- 32* / 44.1 / 48 / 64* / 88.2 / 96 / 176.4 / 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: 8MIC

* This status is set automatically in external synchronization mode as soon as the sample rate is recognized.

The format is set by DIP switch 6 to Professional or Consumer. If AES PRO is chosen, the output level is almost 5V. If CON (Consumer) is chosen, the output signal will have a channel status compatible to SPDIF, and the output level will be reduced to 2V.

Connecting devices with coaxial SPDIF ports to the OctaMic II'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'.*

The OctaMic II 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.

AES/EBU Sync

The input AES 1 (channel 1/2) found on the D-sub connector can not be used for audio with the OctaMic II, but as clock source. 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.

11.2 ADAT Optical

The OctaMic II provides two digital outputs in ADAT optical format. 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 OctaMic II 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.

Up to 96 kHz (Double Speed), the ADAT outputs can be used in parallel to the AES outputs. In Quad Speed mode (128 kHz up to 192 kHz), the ADAT outputs are operated synchronously at Single Speed sample clock, but do not provide any audio data.

The ADAT optical outputs of the ADC Module 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 OctaMic. Carries the channels 1 to 8. When sending a Double Speed signal, this port carries the channels 1 to 4. In Quad Speed mode ADAT MAIN carries an empty but synchronous ADAT signal.

ADAT AUX

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 ADAT AUX carries an empty but synchronous ADAT signal.

12. Word Clock

12.1 Operation and Technical Background

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

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

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



Remember that a digital system can only have one master! If the OctaMic II 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.

The actual end of these problems is offered by the **SteadyClock** technology of the OctaMic II. 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.

12.2 Cabling and Termination

Word clock signals are usually distributed in the form of a network, split with BNC T-adapters and terminated with resistors. We recommend using off-the-shelf BNC cables to connect all devices, as this type of cable is used for most computer networks. Actually you will find all the necessary components (T-adapters, terminators, cables) in most electronics and computer stores. The latter usually carries 50 Ohm components. The 75 Ohm components used for word clock are part of video technology (RG59).

Ideally, the word clock signal is a 5 Volt square wave with the frequency of the sample rate, of which the harmonics go up to far above 500 kHz. To avoid voltage loss and reflections, both the cable itself and the terminating resistor at the end of the chain should have an impedance of 75 Ohm. If the voltage is too low, synchronization will fail. High frequency reflection effects can cause both jitter and sync failure.

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

Ideally all outputs of word clock delivering devices are designed as low impedance types, but all word clock inputs as high impedance types, in order to not weaken the signal on the chain. But there are also negative examples, when the 75 Ohms are built into the device and cannot be switched off. In this case the network load is often 2 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 OctaMic II's word clock input can be high-impedance or terminated internally, ensuring maximum flexibility. If termination is necessary (e.g. because the OctaMic II is the last device in the chain), push the switch at the back (see chapter 9.1).

In case the OctaMic II 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.

User's Guide



OctaMic II

▶ **Technical Reference**

13. Technical Specifications

13.1 Analog

Microphone/Line 1-8

- Input: Neutrik XLR/TRS Combo jack, electronically balanced
- Input impedance: XLR 2 kOhm, TRS 10 kOhm balanced
- Frequency response -0.1 dB: 20 Hz – 100 kHz
- Frequency response -0.3 dB: 10 Hz – 150 kHz
- THD @ 30 dB Gain: < -106 dB, < 0.0005 %
- THD+N @ 30 dB Gain: < -100 dB, < 0.001 %
- Channel separation: > 110 dB
- CMRR 20 Hz – 20 kHz: > 55 dB

- EIN @ 30 dB Gain @ 150 Ohm: 122 dBu
- EIN @ 40 dB Gain @ 150 Ohm: 126 dBu
- EIN @ 50/60 dB Gain @ 150 Ohm: 128 dBu

- EIN @ 30 dB Gain @ 0 Ohm: 122.5 dBu
- EIN @ 40 dB Gain @ 0 Ohm: 128.8 dBu
- EIN @ 50/60 dB Gain @ 0 Ohm: 130.3 dBu

- Gain range: +6 dB up to +60 dB
- Maximum input level XLR, Gain +6 dB: +14 dBu
- Maximum input level XLR, Gain +60 dB: -40 dBu
- Maximum input level TRS, Gain +6 dB: +21 dBu
- Maximum input level TRS, Gain +60 dB: -33 dBu

Line Out 1-8

- Maximum output level: +21 dBu
- Output: 6.3 mm TRS stereo jack, servo-balanced
- Output impedance: 75 Ohm
- Output level switchable Hi Gain / +4 dBu / -10 dBV

AD-Conversion

- Resolution: 24 bit
- Signal to noise ratio (SNR): 110 dB RMS unweighted, 114 dBA
- Frequency response @ 44.1 kHz, -0.5 dB: 5 Hz – 20.6 kHz
- Frequency response @ 96 kHz, -0.5 dB: 5 Hz – 45.3 kHz
- Frequency response @ 192 kHz, -1 dB: 5 Hz - 90 kHz
- THD+N: < -110 dB, < 0.0003 %
- Channel separation: > 110 dB

13.2 Digital Inputs

AES/EBU

- 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)

Word Clock

- BNC, not terminated (10 kOhm)
- Switch for internal termination 75 Ohm
- Automatic Double/Quad Speed detection and internal conversion to Single Speed
- SteadyClock guarantees super low jitter synchronization even in varispeed operation
- Transformer coupled, galvanically isolated input
- Not affected by DC-offsets within the network
- Signal Adaptation Circuit: signal refresh through auto-center and hysteresis
- Overvoltage protection
- Level range: 1.0 Vpp – 5.6 Vpp
- Lock Range: 27 kHz – 200 kHz
- Jitter when synced to input signal: < 1 ns
- Jitter suppression: > 30 dB (2.4 kHz)

13.3 Digital Outputs

AES/EBU

- 4 x, transformer-balanced, galvanically isolated, according to AES3-1992
- Output voltage Professional 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

13.4 Digital

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

13.5 General

- Power supply: Internal switching PSU, 100 - 240 V AC, 20 Watts
- Typical power consumption: 14 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: 2 kg (4.4 lbs)
- Temperature range: +5° up to +50° Celsius (41° F up to 122°F)
- Relative humidity: < 75%, non condensing

13.6 Connector Pinouts

The pinout of the 25-pin D-sub connector uses the widely spread Tascam scheme, which is also used by Digidesign. Out of the four AES inputs only the first one is used by the OctaMic (Sync).

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 labelled 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 analog input and output

The XLR connectors of the analog inputs and outputs are wired according to international standards:

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

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

TRS jacks analog input and output

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

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

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

14. Technical Background

14.1 Terminology

Single Speed

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

Double Speed

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

Quad Speed

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

Single Wire

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

Double Wire

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

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

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

Quad Wire

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

S/MUX

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

S/MUX4

The Quad Wire method 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.

14.2 DS - Double Speed

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

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

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

14.3 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 the OctaMic II does not support this mode at the ADAT outputs.

The AES outputs provide 192 kHz as Single Wire only.

14.4 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).

14.5 SteadyClock

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

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

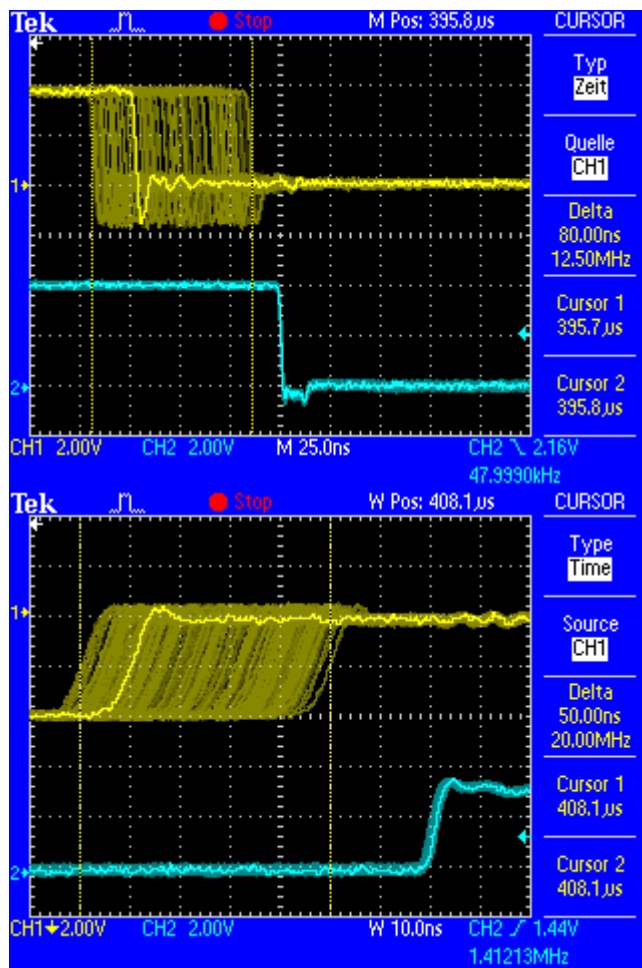
SteadyClock has originally been developed to gain a stable and clean clock from the heavily jittery MADI data signal. The embedded MADI clock suffers from about 80 ns jitter, caused by the time resolution of 125 MHz within the format. Common jitter values for other devices are 5 ns, while a very good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

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

Using the input sources of the OctaMic II, 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-free 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 is also used to clock the digital outputs ADAT and AES/EBU.



15. Block Diagram

