MICROMEGA

Le son de France

White paper

M-One range

M100
M150

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M-One series: A revolutionary all-in-one concept

M-100 is the first item in the new Micromega range. It benefits from the latest advances in digital technology whilst still retaining all the advantages of analogue (Class A/B) audio.

A range of integrated amplifiers and separate components with unequaled connectivity, the M-One series is currently available in two revolutionary models: the M-100 and the M-150. The M-150 differs from the M-100 with the addition of a Power Correction Function (PCF) and the doubling of the audio amplifier components, which significantly improve the power delivery; respectively 2 X 150W at 8 ohms (2 X 300W at 4 ohms) so the M-100 delivers 2 X 100W at 8 ohms (2 X 150W at 4 ohms).

The M-One series is made from a single machined aluminum block and can be used in different positions as the direction of the two displays alters according to the position of the device. The device can be attached to a wall or placed horizontally on a piece of furniture. However the device is placed the display adapts to suit the user.

Another addition to improve user comfort is the size of the icons and letters, which can be changed with the remote control. This means that the user will always be able to read the display whether they are close to or further away from the device.

2 top of the range integrated amplifiers

M100

2 x 100W under 8Ω

M150

2 x 150W under 8Ω

Your One: An amplifier in a color to suit

m.C.F
Micromega Custom Finish

Personalizing your M-One amplifier with Micromega Custom Finish (MCF) means you can choose between different finishes, colours and textures

There are hundreds of different colour, texture and finish combinations. With the help of a network of French artisans we have created our own customization studio. We have the ability to create magnificent pieces which are completely unique.

A brand new listening experience

The electro-acoustic correction system M.A.R.S. corrects any faults in your audio system by taking measurements from a microphone

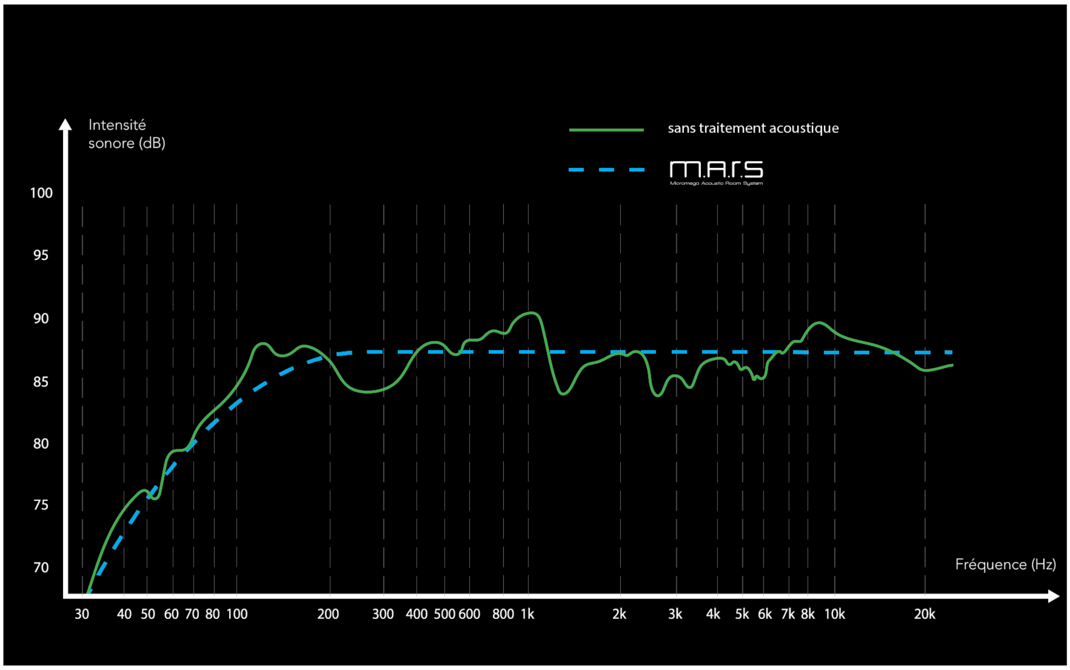
The electro-acoustic correction system M.A.R.S. corrects any faults in your audio system by taking measurements from a microphone placed at the listening point.
This correction system takes into account your whole system: “speakers - room.” It irons out any incidents in the amplitude/frequency response coming from reflections, absorptions or resonances in the listening room. It also corrects imperfections in the speakers.
The sound is transformed, any unpleasant resonance and other acoustic problems caused by your stereo are eliminated. The sound has pinpoint accuracy and the base register has more impact. You can be sure you are getting the best out of your system whilst still preserving the necessary neutrality of true hi-fidelity.

There are two types of electro-acoustic correction on offer:

- Frequency response “Room EQ1” (as standard on the M150 and as an option on the M100)
- Frequency and pulse response ”Room EQ2 (as an option on the M100 and M150)

Micromega has also looked into the integrated headphone output on the front of the M100 and M150 amplifiers. Usually, when you use headphones to listen to music the sound appears to be “in the middle of your ears.” It has lost all perception of width and depth...

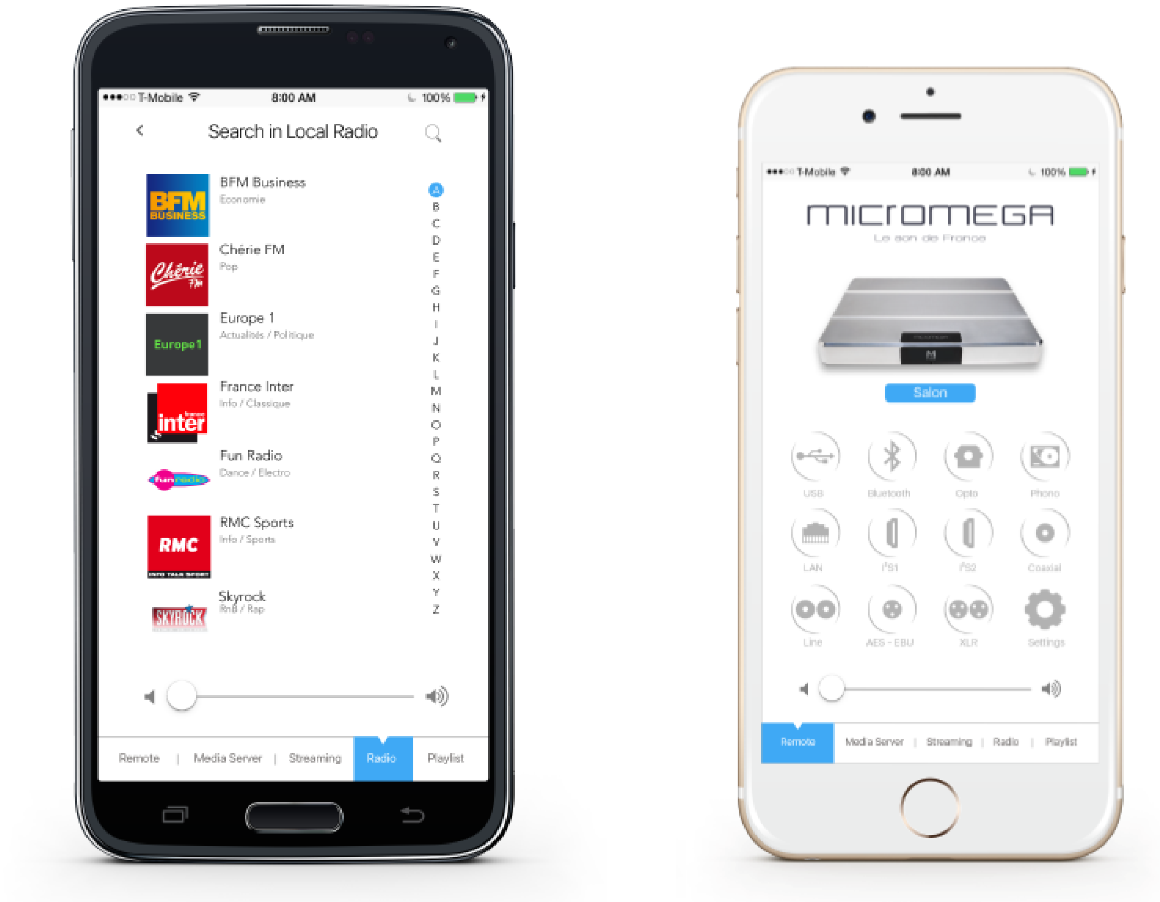
This is where the “Binaural” process comes in. A spatialisation treatment based on HRTF studies (Head Related Transfer Function) has been applied to the headphone output, to reproduce the same listening quality as speakers



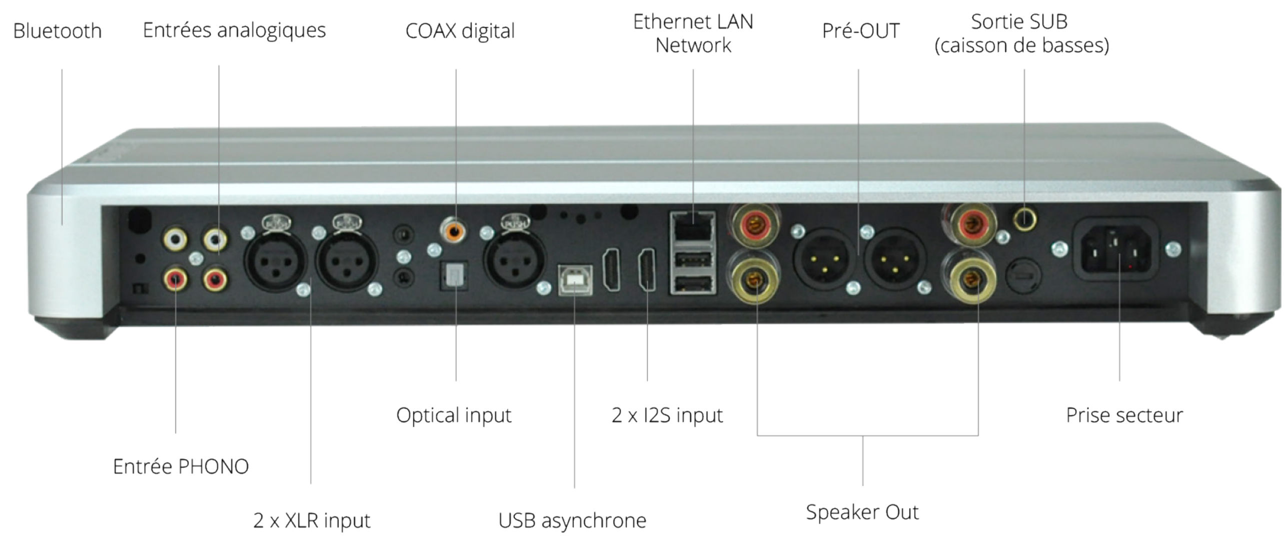
Graphique schématique

A dedicated smartphone app

As well as the remote control, which enables you to select a music source, place the device on stand-by and control the volume, there is also a latest generation smartphone or tablet app (iOS and Android) which offers complete control of the device and media servers over the internet (NAS).



Unrivalled connectivity



M-One works with all kinds of digital and analog inputs, taking into account all current sources and all audio formats.

There are a host of digital inputs:

- A 32 bit class 2 USB audio input (PCM, DSD, DSD/DoP) supporting a DSP XMOS, which is completely insulated from the processing electronics by digital capacitive insulators and optocouplers.
- A SPDIF 32 bit/768KHz coaxial input which is insulated with a transformer.
- A TOSlink 24 bit/192KHz optical digital input
- An AES/EBU 32 bit/ 768KHz input insulated with a transformer.
- A Bluetooth APT-X receiver module which can store up to eight connections with Bluetooth, smartphone or other devices.
- Two I2S inputs on HDMI connectors, to handle future extensions such as wireless point to point or network connections.
- An Ethernet port for networked audio with Micromega app. The completely compatible UPnP - DLNA app can take off stand-by mode via the network. This takes into account the remote control functions of media server management, digital radio and streaming.

M-One handles PCM, DSD et DoP (DSD over PCM) audio stream up to a 32 bit resolution and 768KHz sampling frequency and 11.2 MHz for the DSD.

The device is evolutional and you can update the control programmes by inserting the USB key into one of the two type A USB ports. In the same way, options such as the acoustic correction MARS are activated via this same port.

On a digital level, the data and clock signal are carried out using a CPLD (Complex Programmable Logic Device). This is the nerve centre of the M100. It manages the master clocks with very low phase noise of 45.1584MHz (multiple of 44.1KHz) and 49.1520 (multiple of 48KHz). A SPDIF gateway receiver with SRC ComTrue CT7301 converts all the signals from different sources, SPDIF, DSA and I2S in I2S sent via the CPLD to the Analog Devices SHARC processor of which controls the digital volume on 32bits/word and also MARS (Room and speaker frequency response correction). The signal is re-routed in the CPLD which sends the data in I2S to the digital/analogue converter AKMAK4490EQ. The latter can receive a sample frequency of up to 768KHz. This means the M-One is ready to accept high resolution files via the network and USB ports.

The device has the following analogue inputs:

- A high impedance (1 MΩ) unbalanced line input, so as not to load the external source.
- A high impedance balanced line input.
- A MM/MC low noise switchable phono input.
- A balanced input on a 3.5 mm jack for the microphone which takes the necessary measurements to set up the DSP for the acoustic corrections of the room.

As for outputs, other than the speaker outputs on gold plated terminals designed by Micromega, which accept, banana, fork and heavy gauge cables, there is a balanced L/R preamplifier output on XLR sockets and a Sub mono output with a low pass filter at 400 Hz. Lastly, there is a headphone socket on the front with the option for binaural listening done be the Analog Devices SHARC DSP.

These technology choices have been made for both optimum performance and aesthetics.

The appearance of the M-One and the fact that, for an amplifier, it is very slim, has meant that certain choices had to made, namely with regards to power supply and what sections of power would fit into the casing. Firstly, aside from the display and network module, all of the electrics are hosted on a single card which occupies the whole surface of the device and contains all of the connections. To remove excess heat from the two amplifier channels, a forced convection tunnel, made from extruded aluminum, with an ultra-silent fan (12.8dB SPL at nominal speed) has been added. The rotation speed and the air-flow rate are controlled by the temperature of the thermal heatsink.

In fact the M-One uses two power supplies, one for each channel, which works in a double mono configuration. This means that the two channels are completely independent. The power supply uses LLC* technology for the same tight space problem but also improves the efficacy and quality of audio.

Two resonance power supplies, a double mono structure

Using a resonance power supply has many advantages over a classic transformer with a filter rectification block, aside from smaller dimensions. Firstly, the weight: Each of the M-One 100 power supplies deliver 300W continuously. A classic configuration uses two 300 VA toroidal transformers for a total weight of 8kg, which is already higher than the whole of the M-One. This configuration would be impossible to house with the rest of the electronics in this type of casing. In contrast to other topologies, particularly the LC series, LLC supply means that regulating is as effective whether the device is fully charged or not. MOS transistors are controlled by soft-switching, from zero tension with an almost sinusoidal current wave across the whole control range, which means there are no abrupt wave currents which risk EMC problems and can disturb the electronic supply.

This type of power supply also produces a better dynamic for the stages of amplification and a better transitory response because of its ability to quickly recharge the filtration capacitors with a residual ripple of the power supply, which is almost nothing. The resonance power supply works on a frequency range between 90KHz and 120KHz, so with the rectifier components around 200KHz and beyond, that's six times over the audio range. With the reject effect of amplification stages in class AB you don't have to worry about any modulation or intermodulation.

As the capacitors are recharged 2000 times faster, you do not need a large energy storage capacity even when the device is fully charged. This considerably decreases the space occupied by the filtration capacitors. In fact, the components used should be of excellent quality and the M-One has the best available. With the reject effect of a AB class amplification configuration, a very soft residual noise can be obtained and a total restitution of micro-information of the audio signal can be completely restored. In short this type of power supply has great efficiency, in the range of 95% when fully charged and stand-by consumption is less than 1W including auxiliary power.

*LLC: so called resonant power supply uses a high frequency converter which presents strong leakage inductance caused by a loose connection because of the way it is constructed. The magnetising inductance and the leakage inductance (Where the LL in LLC comes from) produces a parallel resonance (LM) with the tuning capacity and a resonance series (Lf). On the range between the two types of resonance, global impedance is inductive with a slope and a power factor protection which depends on the load. The control loop means you can work between the two resonances and beyond the resonance series (empty) whilst always having the presence on an inductive impedance to the primary and consequently zero tension switching of the switching devices and a triangular (beyond Fr series) or almost sinusoidal secondary current (between Fr series and FR parallel). This type of supply achieves optimum regulation to the same level when empty or fully charged, unlike other types of resonance power supply.

Class A/B power levels

A/B
CLASS

For Micromega, at equivalent power, a good class AB amplifier offers the best characteristics and better sound restitution than a class D amplifier, at the detriment of efficacy, in other words, the power delivered compared to power absorbed.

In contrast to class AB, with a Class D amplifier the rejection of power is very low. So it is almost inevitable that HF switching residues will be recovered and these create intermodulation because IMD (Intermodulation Distortion) is less effective on a class D amp than a well constructed class AB amp.

With a class D amp you have to use an output filter to smooth (integrate) the output signals and this filter is never completely adapted to the different loads of the speakers. If it is optimised for 8Ω, then it is of no use for 4Ω because a compromise will have to be reached. What is more, the speakers are a complex load and not pure resistance. The bandwidth of a class D amplifier is less stretched and consequently, the phase turns faster on the useful audio band, which gives a less effective impulse response.

Finally, we cannot totally eliminate odd order (harmonics 3 and 5, Total Harmonic Distortion) that is inherent in down time and obligatorily introduced between phases of switching between power devices, which always gives a harder sounds.
In retrospect a class D amplifier gives better efficacy, so less thermal dissipation is required and there is a better weight to load ratio, essentially because of the limited size of the dissipaters.

With a class AB, it is then necessary to have the correctly size the thermal convection heatsink, as any difference between the power absorbed and power returned is transformed into heat. The average yield is around 50%. M-One uses an extruded, forced convection cooling tunnel with a ultra-quiet fan (no ball-bearings, the axis is held in place using magnetic levitation); this tunnel crosses through the device from one side to the other. This set up means the dissipater can be held within the low height of the M-One which would otherwise have been impossible. It is also coupled thermally to the aluminum casing, which increases the dissipation capacity.

The other problem with the class AB amp is in the stability of polarisation of the power stage. On the M-One the power transistors have a diode at the heart of the body, which exactly copy the voltage variations of the base-emitter junction depending on the temperature (Thermal track ON Semiconductor). In this way, polarisation does not drift with the power delivered and the rising temperature thus gives a better distortion figure.

A totally balanced audio analogue path

From the digital-to-analogue converter (DAC) the whole analogue distribution is carried out in balanced liaison. The conversion to unbalanced happens just before the amplification stages. This means the fluctuations in common mode are better rejected and the signal/noise relationship is optimised. On the balanced output, the signal is always transported as balanced from the DAC.

This structure is also used for the un-balanced analogue inputs which are balanced before arriving at the DAC.