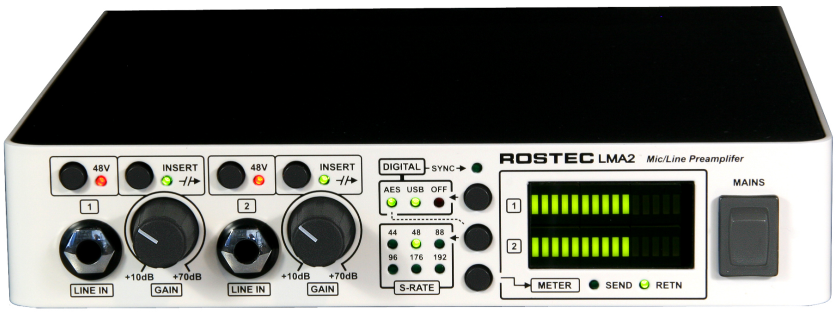
ROSTEC LMA2 Microphone and Line Preamplifier



ROSTEC LMA2

High Performance Microphone and Line Preamplifier

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Features

- **2 microphone inputs via the 25 pin D-sub female connector at the back panel**
- **2 high impedance instrument inputs at the front panel**
- **2 analog line outputs at the back panel**
- **2 analog insert points at the back panel, insert between analog preamplifier and AD converter (USB/AES/SPDIF outputs)**
- **Insert points can be selected on/off at the front**
- **All switching is done by hermetically sealed relays with gold contacts**
- **Full wave true peak meter, range 40 dB, attack 0.2 msec, release 5 sec**
- **Full balanced internal analog architecture**
- **Built-in +48 Volt Phantom Power Supply**
- **Smooth gain adjustment from +10 dB to + 70 dB by potentiometer, one turn - no clicks**
- **USB 2.0 Class Compliant Digital Interface**
- **Transformer balanced AES3 110 ohm output.**
- **Unbalanced SPDIF 75 ohm output**
- **AD Resolution: 24 bits @ 44.1k, 48k, 88.2k, 96k, 176.4k, 192k**
- **USB latency down to 1.5 msec, dependent on the host computers capabilities**
- **AES latency: down to 0.05 msec, depending on sampling frequency**
- **Analog input noise -134 dBu (A weighted)**
- **+30 dBu mic/line input headroom**
- **+30 dBu max analog output level**
- **Analog outputs and Insert points nominal levels are +18 dBu at DFS (EBU standard)**
- **Analog frequency response is 5 Hz - 200 kHz +/- 0.1 dB**
- **Analog crosstalk -120 dB @ 1kHz, input terminated by 150 ohm**
- **Digital crosstalk -110 dB @ 1 kHz**
- **THD+N: 0.00035 %, THD: 0.00014 %**
- **Linear Low Noise Power Supply**
- **Mains voltage range 180 - 264 VAC 50-60 Hz (115 VAC version available by request)**
- **Power consumption 8 Watts**
- **All digital outputs are ESD protected to 23 kV 15 A surge (IEC61000-4-2 and IEC61000-4-5)**
- **Analog in- and outputs are ESD protected to 2000V, human body model**
- **Built for real life! Sturdy steel casing, magnetically and electrically screened**

General description

The LMA2 is a 2 channel ultra low noise and ultra low distortion mic/line preamplifier designed with the professional sound engineer in mind.
The design is based on a very stringent philosophy, meaning the shortest possible signal path and the highest possible quality components.
The input circuit is a true class A differential gain stage, providing superior sound quality and high speed for exceptional audio performance.
The circuit has a transfer characteristic that resembles that of vacuum tube triodes, giving the unit a natural, relaxed and open sound, yet it maintains an extremely fast and totally precise impulse response.

The unusual large input headroom of +30 dBu enables the amp to handle fast transient and large dynamic sound pressure changes with ease.
The circuit is designed so input clipping cannot be experienced. The input circuit has a clipping limit at +30 dBu, and because there is always at least +10 dB of gain in the signal path, the output (or the connected equipment) will simply always clip before the input.

The analog architecture is fully balanced throughout the unit, which means that the signals between the various circuits are routed as a positive and a negative signal, not the standard way of using signal and ground. Ground is not used to transfer analog audio signals at all. This architecture keeps the audio path free from non-linear distortion from currents running in the ground plane and from “non musical” signals from external electrical fields, power supply noise, crosstalk from other channels etc.

The LMA2 is constructed by using modern day's cutting edge technology, but the design philosophy is inspired by some of the very best preamps that have been manufactured over the last 50 years.

Microphone inputs and instrument/line inputs

The amp has 2 separate and identical analog channels. Each channel has two inputs, a microphone input and a high impedance (High-z) line/instrument input.
The 2 microphone inputs can be accessed at the 25 pin D-SUB female connector at the back panel.
The 2 instrument/line inputs are placed at the front panel for easy access and use 1/4" standard Jack connectors.

Switching between the line input and the microphone input is automatic. When no Jack is plugged into the front connector, the microphone input is active.

When a Jack is plugged into the front connector, the unit automatically disables the microphone input at the back, activates the line input at the front, and switches to high impedance mode. The high impedance of the line input (High-z) is intended for instruments, like guitar or bass, but it is equally well suited for line level equipment, such as keyboards and the like.
Observe that the line input at the front is balanced, but can equally well be used by unbalanced and balanced sources (see operational description, page 11).

Analog Outputs and insert points

Each channel has a buffered balanced direct analog line output for use with external analog equipment.
Each channel also features a balanced insert point, which serves as a breaking point between the analog section and the digital section. It is intended for inserting external analog equipment in the signal chain, like for example your beloved vintage compressor, equalizer, effect unit etc.
The individual insert points can be opened and closed by the switches on the front panel. When the insert point is opened, it also doubles as a direct line input to the analog to digital converter (via insert return).
The direct line outputs and the insert points (insert send and insert return) can be accessed at the 25 pin D-SUB female connectors at the back panel.
All output buffers are floating industrial grade output buffers. They are equally well able to handle balanced as well as unbalanced loads.

USB outputs and AES/SPDIF outputs

The LMA2 features a high-end 2-channel 24 bit 192 kHz AD converter with impressive specifications.
The AD analog input circuits are state of the art using high quality top range components, and all analog filters are of minimum phase design with strictly controlled impulse responses.
Each of the two AD analog input channels features a soft clipper with a threshold just a little bit above digital clipping. This means that you can ride the converter reasonable hard and continuous at the clipping level without getting the ugly sound break-up so well known from standard digital interfaces and sound cards.
Input clipping of the LMA2 mic/line inputs is completely irrelevant, because the clipping level of the input circuit is 12 dB higher than the digital clipping level of the AD converter.

Analog input clipping is simply not an issue with LMA2. (See technical section for clipping characteristics)

The AD converter feeds the following three digital outputs simultaneously:

1.
A USB 2.0 Class Compliant streaming audio interface. via a USB type B connector, the most physical robust of all USB connectors. The USB connection features a bit-perfect transmission protocol, and is class compliant and compatible with USB 2.0, 3.0, 3.1, 3.2 and USB-C.
The USB interface supports Core-Audio on MAC OS X systems. When a connection to a MAC host computer is made, 2 analog input channels and 2 output channels show up on the MIDI-setup panel and are immediately available as system resources for any DAW.
The 2 outputs can transfer digital audio directly from USB to AES/SPDIF (via the DAW).
Windows is not supported. The Windows operating system is NOT recommended to handle professional streaming audio equipment via USB.

2.
A transformer balanced AES3 output via the 9 pin D-SUB female connector at the back panel. The output format follows all the relevant Audio Engineering Society recommendations.
The AES output is transformer balanced, impedance 110 ohm, 6.6 Volts no load, and 3.3 Volts into 110 ohm.

3.
An unbalanced SPDIF output via the 9 pin DSUB female connector at the back panel. The output format follows all the relevant Audio Engineering Society recommendations.
The SPDIF output is unbalanced, impedance 75 ohms, 1 Volt no load, 0.5 V into 75 ohm.
When connecting 2 pins on the 9 pin D-SUB connector, the SPDIF format can be switched between consumer format and professional format (default)
Pin 9 and 5 connected: Consumer format.
Pin 9 and 5 not connected: Professional format

Clock sync input and output

The LMA2 uses an internal high performance master clock generator to run all the necessary clocks for the various digital circuits. It has an input and an output for clock synchronization on the standard BNC connectors at the back.

When the LMA2 is connected to a host computer via USB, the sampling frequency is controlled by the operating system, or it is selected from within the DAW. Manual control of the sampling frequency is disabled when the USB connection is used.

When no USB connection is used, the sampling frequency can be selected manually by the pushbutton at the front.
However, when an incoming word clock is present at the BNC clock input connector, the LMA2 automatically switches to the corresponding sampling frequency, and manual control of the sampling frequency is disabled.

The LMA2 automatically detects when an incoming word clock is present at the BNC connector at the back panel, and the internal generator locks on to it immediately.
The internal generator softly glides from the internal crystal reference to the external clock reference without any jumps or disruptions of the internal clock signals
It achieves lock in typically less than 0.3 seconds, and if the incoming sync is lost, it softly glides back to the internal crystal reference again. There are no gaps or interruption of the internal clocks or any audio data.
Shorter gaps in the incoming sync are efficiently absorbed that way. The soft gliding back and forth is sufficiently gentle and well damped in order for the digital circuits to handle this without any degradation of the audio signal.

Further, the generator has an extensive ability to clean op a malformed and distorted incoming clock.
The input uses a high-speed comparator with hysteresis and a "sweet spot" detector, which performs an accurate auto-slicing of the input.
This means that the circuit automatically chooses the most useful part of the input signal, thus being able to clean-up and reconstruct a ringing and noisy input clock into a perfectly shaped clock for internal use

An input clock with high level of jitter gets the treatment too. The internal PLL uses a multi pole filter network to make it largely immune to incoming jitter.
Put in another way, when the timing of the leading edge of the incoming clock varies with time (this is what jitter is!) the crystal oscillator won't follow these fluctuations, but chooses the average position of the leading edge of the incoming clock. The result is a cleaned-up reference clock with typically less than 0.1 nsec RMS jitter.
This solid jitter performance and high stability of the clock generator provide an ideal environment for the AD converter, the USB state machine and the AES/SPDIF transmitter circuit.
This is one of the basic reasons for the exceptional sonic quality of the LMA2 digital interface.

The meter

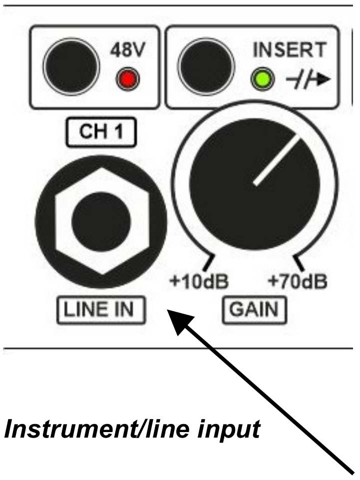
The meter is a true peak meter with fast attack and slow release, with a correct professional ballistics profile (not like the rubbery and sloppy meters on a typical DAW).
The switch on the front panel selects two measuring points: SEND measures the insert point send level (same as analog out), and RTN measures the insert point return level. Note that the meter is a measuring device, it does not route any signals. The meter provides the user with a trustworthy indication of levels and clipping of the digital converters, microphone amplifiers and the inserted analog equipment

Power Supply

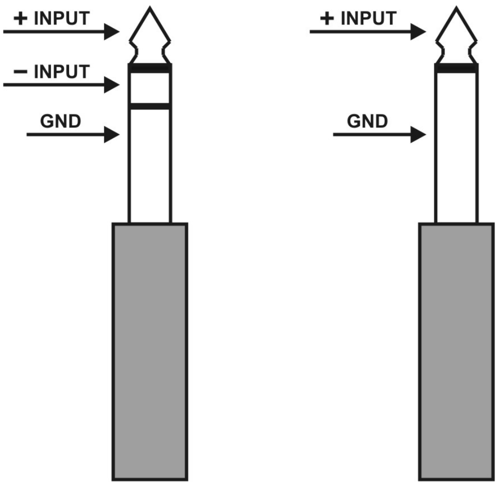
The power supply is pure analog, with an oversized toroidal transformer and low noise linear regulators.
This configuration creates an electromagnetic quiet environment, free from the typical radiation pollution from switch-mode power supplies
A switch mode power supply generates strong repetitive electromagnetic pulses that travel through air and sharp current pulses that travel through the ground system.
When this pollution hit the analog circuitry, it disrupt the smooth operation of the circuitry by pressing the amplifiers into **slew-rate-mode** momentarily

Slew rate mode, is when an amplifier is presented with a signal that moves faster that the amplifiers maximum speed capability. It then tries to “slew” as fast as it can, to cope with the signal. When the amplifier is in this mode, it cannot process any further information; it is in fact blocked from reproducing incoming audio signals.
This happens in short durations, when the pulses from a switch mode power supply hit the circuit.
When this electromagnetic pollution hits the analog circuit 60.000 times a second, the analog circuit looses, in small intervals, the ability to reproduce audio.
This is in fact the main reason why audio products with switch mode power supplies usually sound harsh, flat and lifeless, with a degraded ability to process details and depth in the audio.
LMA2 maintains a clean electromagnetic environment in the box, and the reward is a natural, musical, relaxed and open sound.

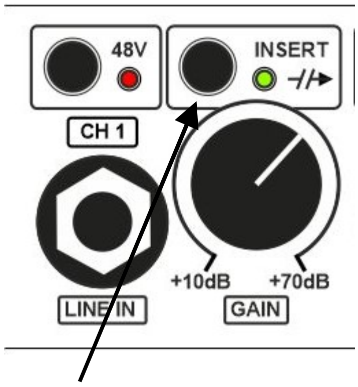
Operational description



Instrument/line input



Each channel has a ¼" Jack instrument/line input on the front panel. The input is for use with instruments (like guitar, bass etc) and it can be used in a balanced as well as an unbalanced configuration.
When a Jack is plugged-in, the input circuit automatically disconnects the mic input, switches to high impedance mode (Hi-Z) and activates the line input.
Plugging in a mono Jack enables the normal unbalanced input configuration.
Plugging in a stereo Jack enables the balanced input configuration.
For further information about using the balanced configuration see ***Remote ground connection***

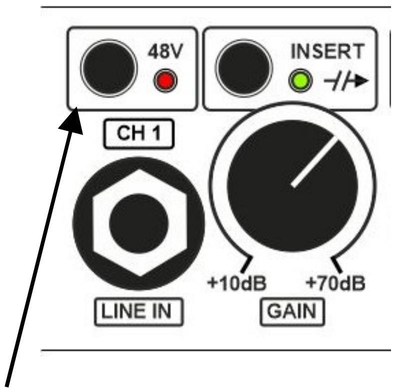


Insert point switch

The toggle switch controls the bypass function of the insert point. The insert point exists electrically between the analog output of the preamp and the input of the digital circuit.

When the LED is ON, the signal chain between the preamp and the digital circuit is “open” which means that external equipment can then be inserted between the insert send and the insert return at the back panel

When the LED is OFF, the analog signal is routed directly from the preamp to the digital circuit. The insert return/direct input is then inactive.
The insert send is not affected, and is always active, and can be used as an extra analog line output.



+48V Phantom Voltage

The toggle switch turns the Phantom Voltage at the mic input on and off. The LED indicates the status.

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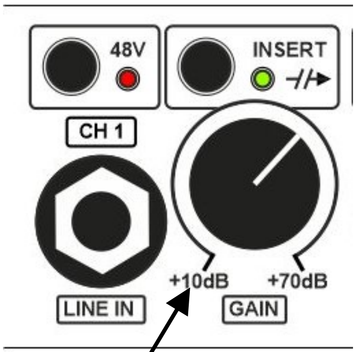
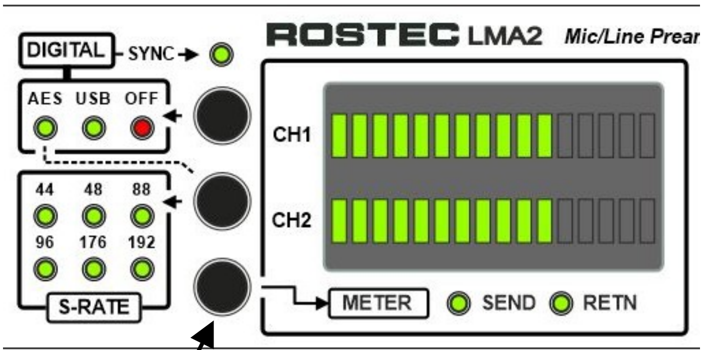


Input gain adjust

Turning the knob controls the input gain from minimum to maximum in one smooth movement. No clicks and no jumps. It works for line input as well as microphone input.
The gain range is +10 to +70 dB from input to the analog line output on the D-SUB connector at the back.

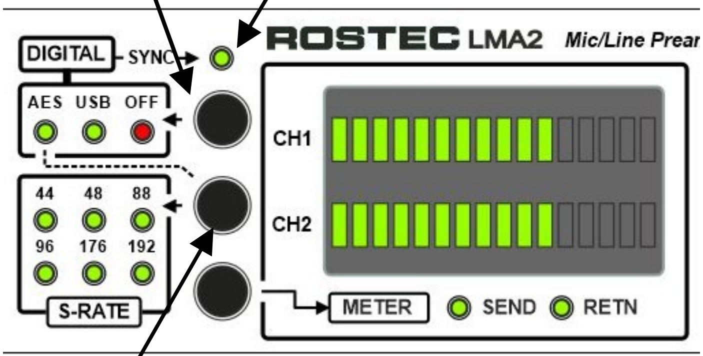


Meter

The button selects the meters measurement points. The meter measures send and return signals at the insert points. *Note that this switch does not change or route any signals in the signal chain!*
SEND: Analog signals sent from the input circuit to external equipment (same signal as analog direct out).
RETURN: Analog signals received from the external equipment to the digital converter circuit.

The button selects the digital functions:
OFF led on: All digital functions are disabled.
USB led on: The unit is communicating via USB.
AES led on: AES/SPDIF from AD converter circuit.

Indicates that the LMA2 is locked to external word clock.



Digital controls and indicators

While USB is **<u>active,</u>** the button selects the AES/SPDIF source:
1: **AES led on**. AES/SPDIF from AD converter circuit
2: **AES led off**. AES/SPDIF from DAW

While USB is **<u>not active</u>**, the button selects the sample rate.
The sample rate is shown by the 6 LED indicators:
44, 48, 88.2, 96, 176.4, and 192 kHz.

Front panel quick guide

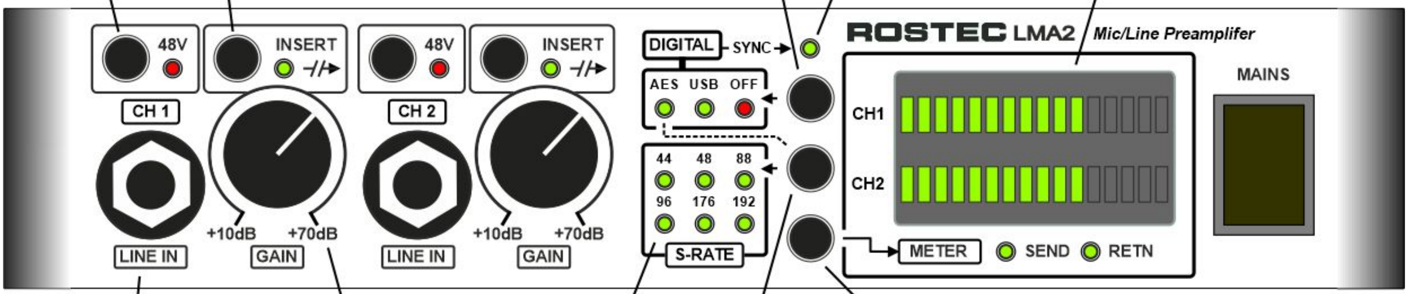
The button selects the digital functions:
OFF led on: All digital functions are disabled.
USB led on: The unit is communicating with a DAW via USB.
AES led on: Analog signal is converted to AES/SPDIF.
AES led off while USB led is on: AES/SPDIF is routed from DAW.

Activates the insert point between the analog circuit and the digital circuit.

Activates the +48 V phantom voltage for the microphone input.

Indicates that the unit is locked to external word clock.

Fast full wave peak meter.
Accuracy at digital clipping 0.2 dB.
Range 40 dB.
Attack time 0.2 msec.
Release time 5 sec.



Gain control for line-in and microphone input.

Sample rate.

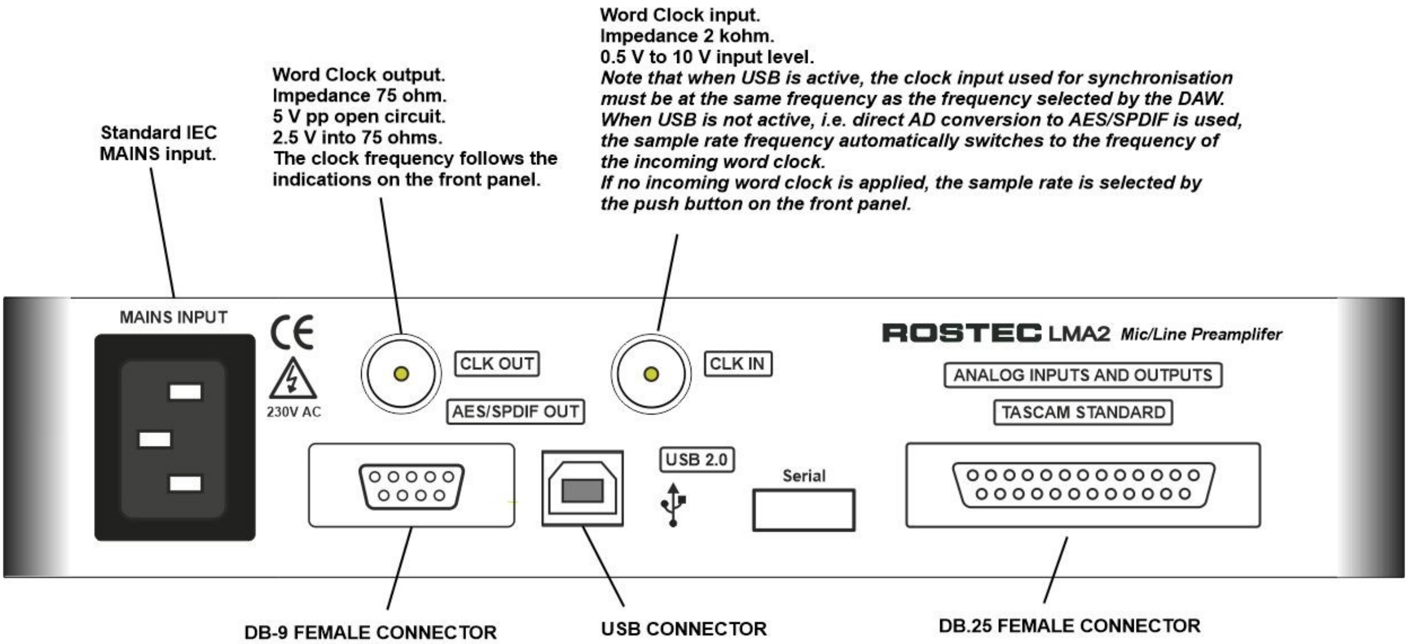
The button selects the meter source. Note that this is not a switching or routing function. It's just for measurement.
SEND: Send to input of external equipment (same as main out).
RETN: Receive from output of external equipment.

High impedance line-in for instruments. Plugging in a jack connector automatically disables the microphone input at the back panel DB-25 connector.
Note: The jack input is balanced, If a balanced jack connector is used, make sure that the ring is connected to ground or to the negative pole at the signal source.
If a standard unbalanced jack connector is used, no considerations are necessary. Just plug it in.

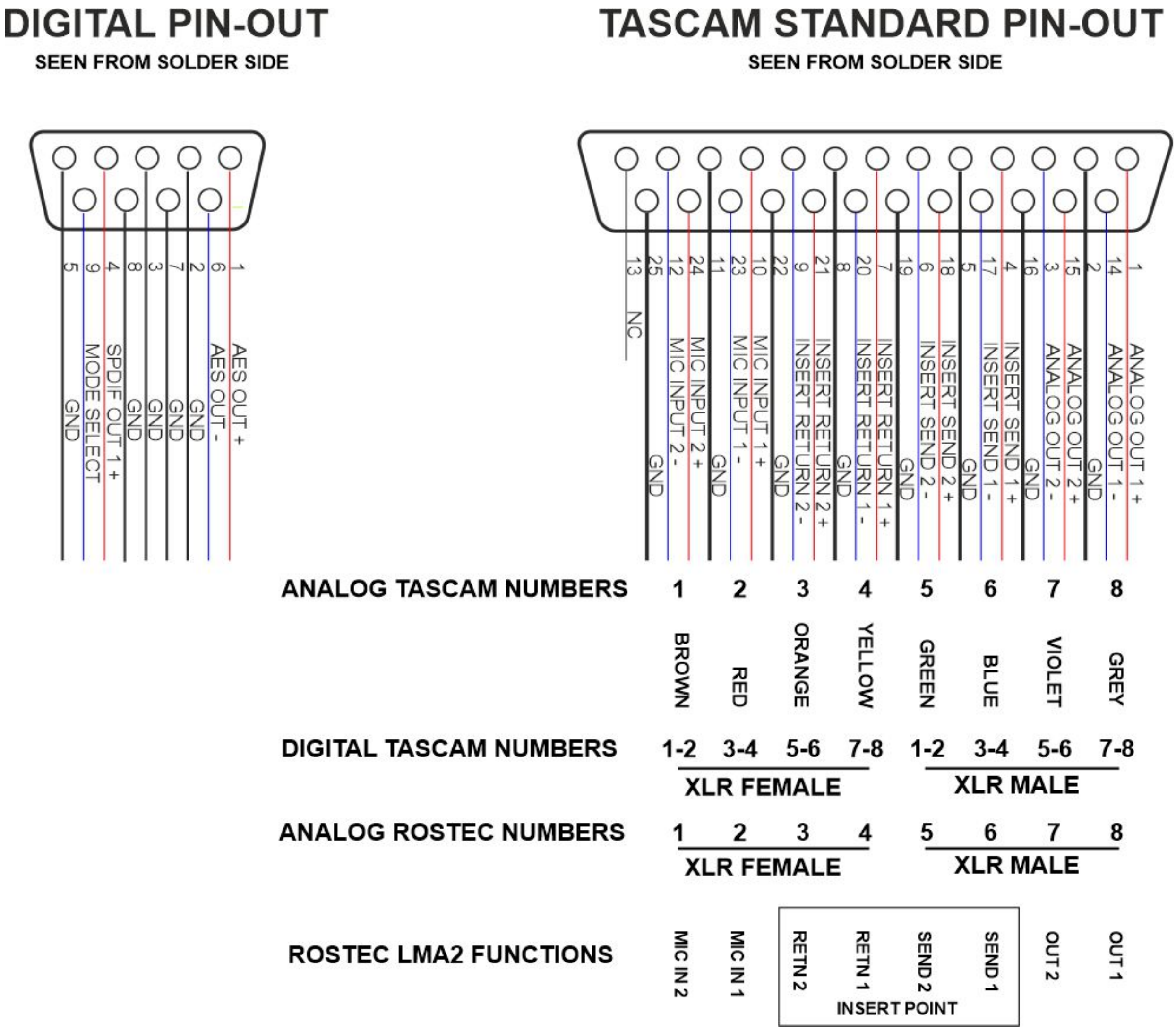
While USB is active, the button selects the AES/SPDIF source:
1: AES led on. AES/SPDIF from converted analog signal.
2: AES led off. AES/SPDIF from DAW
While USB is active, the sample rate is always controlled by the DAW.

2: While USB is not active, the button selects the sample rate.

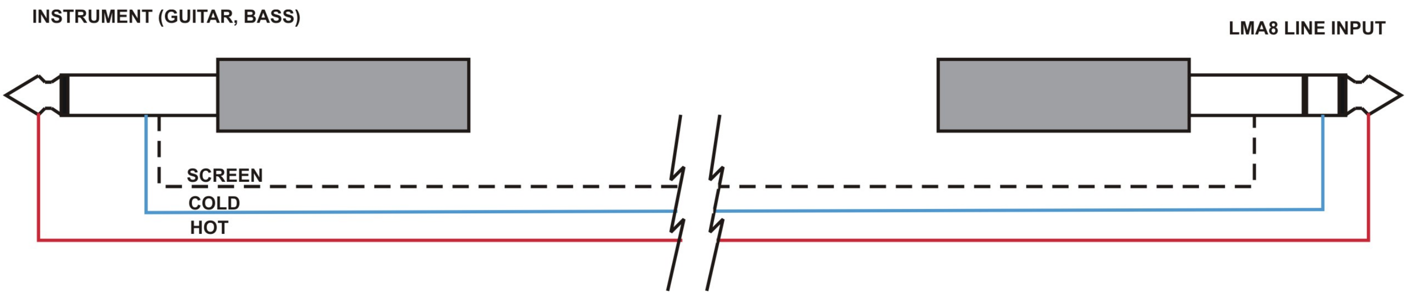
Back panel quick guide

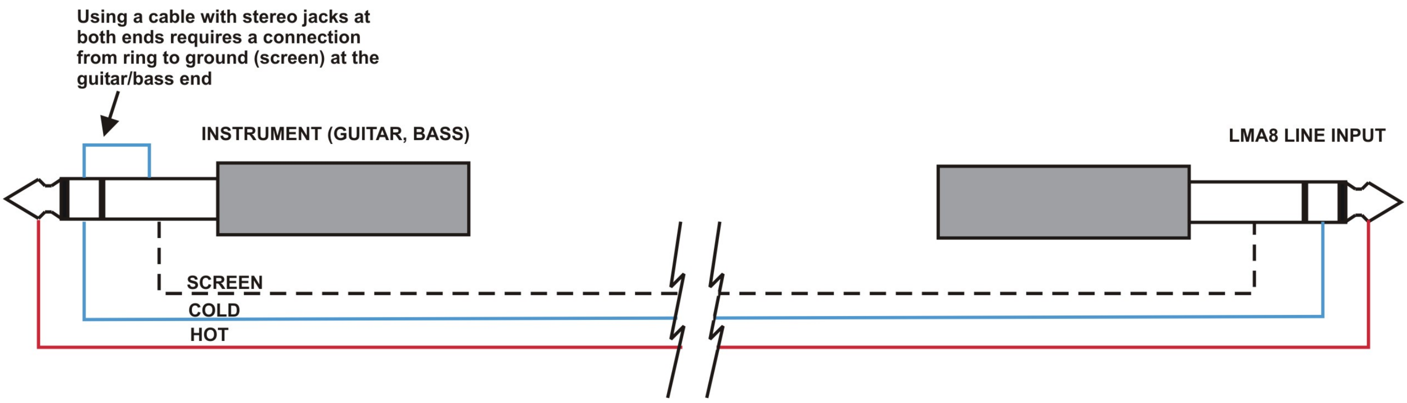


Back panel connector pin-out



Remote ground connection for instrument/line input





The balanced instrument/line input of the LMA2 opens up the unique possibility to use a pseudo balanced configuration with instruments like guitar, bass etc. The configuration will cancel out a large portion of the electrical noise induced in the cables. This is quite advantageous when using long cables on stage or in an electrically noisy environment near light dimmers or near high power mains installations.

A guitar or bass hook-up is a high impedance system, and as such, it is very sensitive to electrical interference. Using a balanced cable instead of a single core cable, introduces one additional wire to "receive" the environmental noise. This additional noise signal is fed into the balanced input of the LMA2 at the negative input terminal, canceling out the noise signal at the positive terminal without affecting the sound signal from the instrument.

The configuration will in most cases give a substantial noise reduction, depending on the electrical characteristics of the instrument pick-up. The two signal lines are terminated by different impedances at the instrument and at the preamp input, so a total noise cancellation will not take place. As a minimum, a noise reduction of at least 10 - 15 dB should be expected.

Note that the noise from the instrument pick-up is not reduced. Only the noise induced into the cable is affected.

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Mechanical and electrical specifications

Analog circuit:
Mic input impedance: 6 kohm
Line input impedance: 1 Mohm/60 pF
Max input level balanced, mic or line: +30 dBu
Max input level unbalanced, line: +24 dBu
Max input level balanced, direct input: +30 dBu
Max input level unbalanced, direct input: +24 dBu
Max output level, balanced output: +30 dBu
Gain, input to balanced output: +10 dB to +70 dB
Input noise: -134 dBu, A weighted, 20 Hz - 20 kHz
Input noise: -131 dBu, unweighted
Line output buffer noise: 118 dB RMS, A weighted, 20 Hz – 20
Hum residue from power supply, 1st, 2nd, 3rd and 4th harmonic: less than -136 dB referenced to 0dBFS.

Analog frequency response: 5 Hz - 200 kHz, +/- 0.1 dB
Crosstalk: -120 dB @ 1 kHz, input terminated by 150 ohm
Nominal analog output level +18 dBu (EBU standard)
Nominal insert point level +18 dBu (EBU standard)

<u>Distortion + noise, classical analysis:</u>
THD+N 0.00035 % @ 1 kHz, 10 dB gain
THD+N 0.00075 % @ 1 kHz, 30 dB gain

<u>Distortion, FFT analysis:</u>
THD 0.00014 % @1 kHz, 10 dB gain
THD 0.00014 % @1 kHz, 20 dB gain
THD 0.00015 % @1 kHz, 30 dB gain
THD 0.00024 % @1 kHz, 40 dB gain
THD 0.00046 % @1 kHz, 50 dB gain
THD 0.00092 % @1 kHz, 60 dB gain

Meter:
Full wave true peak meter, range 40 dB, attack 0.2 msec, release 5 sec
Clipping accuracy 0.2 dB, @+18 dBu analog level

Digital interface:
USB 2.0 Digital interface, Class Compliant, TYPE B connector
USB latency: down to 1.5 msec. This is highly dependent on the host computer
The USB 2.0 connection is compatible with Mac OS X 10.6.8 or later. Windows is NOT supported.
AES output, .transformer balanced, impedance 110 ohm, 6.6 Volts no load, and 3.3 Volts into 110 ohm.
SPDIF output, unbalanced, impedance 75 ohms, 1 Volt no load, 0.5 V into 75 ohm.
AES latency: down to 0.05 msec, depending on sampling frequency
AD Resolution: 24 bits.
Sampling frequencies: 44.1k, 48k, 88.2k, 96k, 176.4k, and 192k.
Digital Interchannel gain mismatch: 0.1 dB
Digital crosstalk -110 dB @ 1 kHz.
Linearity inside passband (passband ripple): 0.035 dB.
Frequency response: 2nd order analog anti-aliasing filters at 1 Hz - 120 kHz +/- 0.2 dB. Brick-wall digital filters at 0.5 x Fs
THD+N: 0.00038 % (-108 dB) at 1 kHz (classical analysis)
THD: 0.00014 % (-116 dB) at 1 kHz (see FFT distortion analysis).

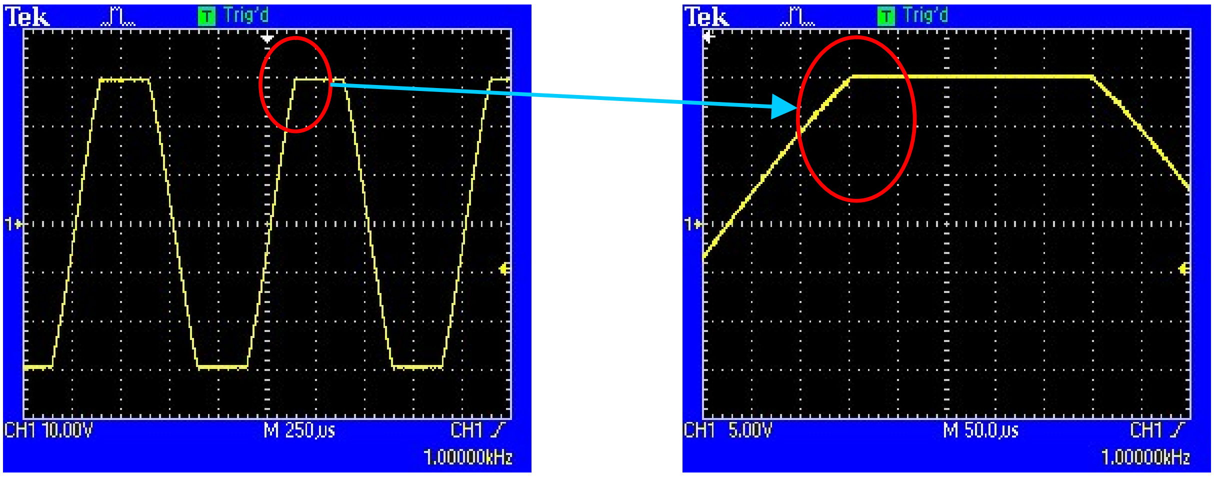
Clock system:
Word clock input, BNC, unbalanced, unterminated (1.5 kohm), min. 0.2 Vpp, max. 10 Vpp.
Word clock output, BNC, unbalanced, impedance 75 ohm, 5 Vpp no load, 2.5 Vpp into 75 ohm.
Word clock frequencies: 44.1k, 48k, 88.2k, 98k, 176.4k, 192k.
Word clock output jitter: 50 ps rms.
AES/SPDIF data jitter: 50 ps rms.
Internal voltage controlled crystal oscillator, jitter 2 ps RMS
Temperature stability +/-2 ppm from 0 degC to +70 degC.
Ageing 1 ppm per year.
Input capture range: +/- 100 ppm.
Lock time approx: 0.2 - 0.4 sec.

Power supply:
Linear analog, passively cooled, high power, very low noise.
Thermally and overload protected.
Mains voltage: Nominal 230VAC; 180 VAC - 264 VAC 50-60 Hz (115 VAC version available by request).
Power consumption: 8 Watts.

General:
Dimension: Width 210 mm, height 42 mm, depth 210 mm
Weight: 2.0 kg
Power: 180 - 253 VAC, 50-60 Hz, 15 Watts
Digital inputs and outputs are ESD protected to 23 kV 15 A surge (IEC61000-4-2 and IEC61000-4-5).
Analog inputs are ESD protected to 2000V, human body model.

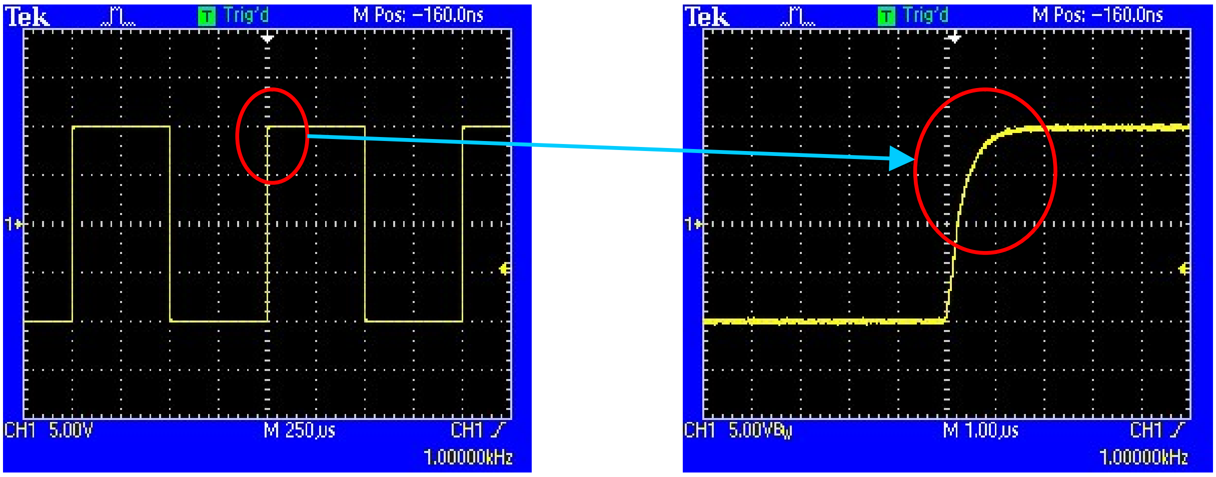
Technical section

Clipping characteristics



The LMA2 analog circuit shows excellent clipping characteristics. When the amp clips, there is no bouncing, ringing, pumping etc. There is just regular and plain clipping. A closer look at the cutting edge of the clip can be seen on the right picture. Notice that there is no overshoot and no recovery delay. There is only instant clipping and instant release.
This kind of clipping is barely audible on short transients. Note that the output level is +30 dBu, so most equipment connected to the amp have already gone into severe clipping at this level.
The input clipping always occurs at a 10 dB higher level than the output clipping, thus the input clipping is always masked by the output clipping. Input clipping cannot be transferred to the output.

Impulse response



The above snapshots show the ideal step impulse response of the LMA2 analog circuit. When subjected to steep transients (in this case a 1 kHz square wave with 5 nsec rise time), there is no ringing and no overshoot. There is only total control. The amp does NOT produce any signal itself when subjected to transient material. It does not add anything. It stays true to the source!
A closer look at the leading edge of the step impulse can be seen on the right picture. There is absolutely no ringing or overshoot. Also, the amplitude of the step on the picture equals approximately +20 dBu audio level, yet the circuit does NOT go into slew rate mode. The curve remains a true exponential.
The output voltage swings 20 Volts in less than 1 usec. Rise time (10/90 % of the amplitude) is approx 600 nsec

Unwanted HF feedback and oscillation in studio installations

It may seem risky to install a piece of equipment witch exhibits such fast speed and high gain at the same time in a studio environment. Normally, when the input signal path and output signal path of a amplifier with high gain and high frequency response passes too close to each other, high frequency oscillation may occur.
This usually happens if the signals are routed through a mixing console of inferior quality, a bad patch-bay or through a piece of equipment with bad crosstalk performance.

But not to worry; LMA8 has a mechanism in place to prevent such unwanted HF feedback.

HF oscillation in complex studio installations is prone to occur when both gain and frequency response is high. The LMA8 uses a well established technique that employs a "constant gain/bandwidth product" function. In short; it works by reducing the upper frequency response in a controlled manner when the gain is increased.

The numbers below illustrates this:

Gain: +10 dB	Upper frequency response (-0.1 dB)	200 kHz
Gain: +20 dB	Upper frequency response (-0.1 dB)	153 kHz
Gain: +30 dB	Upper frequency response (-0.1 dB)	130 kHz
Gain: +40 dB	Upper frequency response (-0.1 dB)	90 kHz
Gain: +50 dB	Upper frequency response (-0.1 dB)	47 kHz
Gain: +60 dB	Upper frequency response (-0.1 dB)	24 kHz

The change of the frequency response is not audible and it does not affect the internal gain structure, noise figure or distortion figure.

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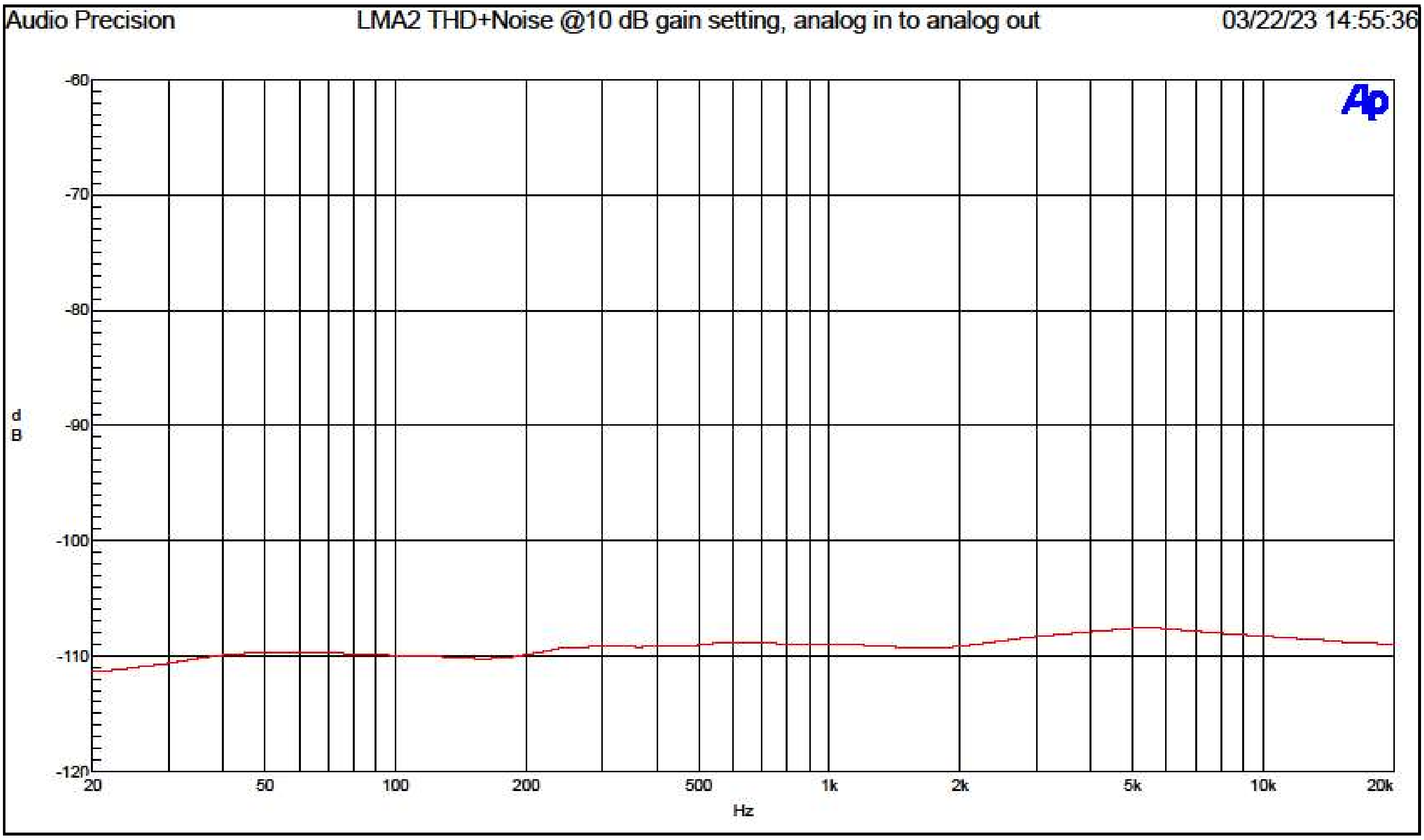
Classical distortion analysis

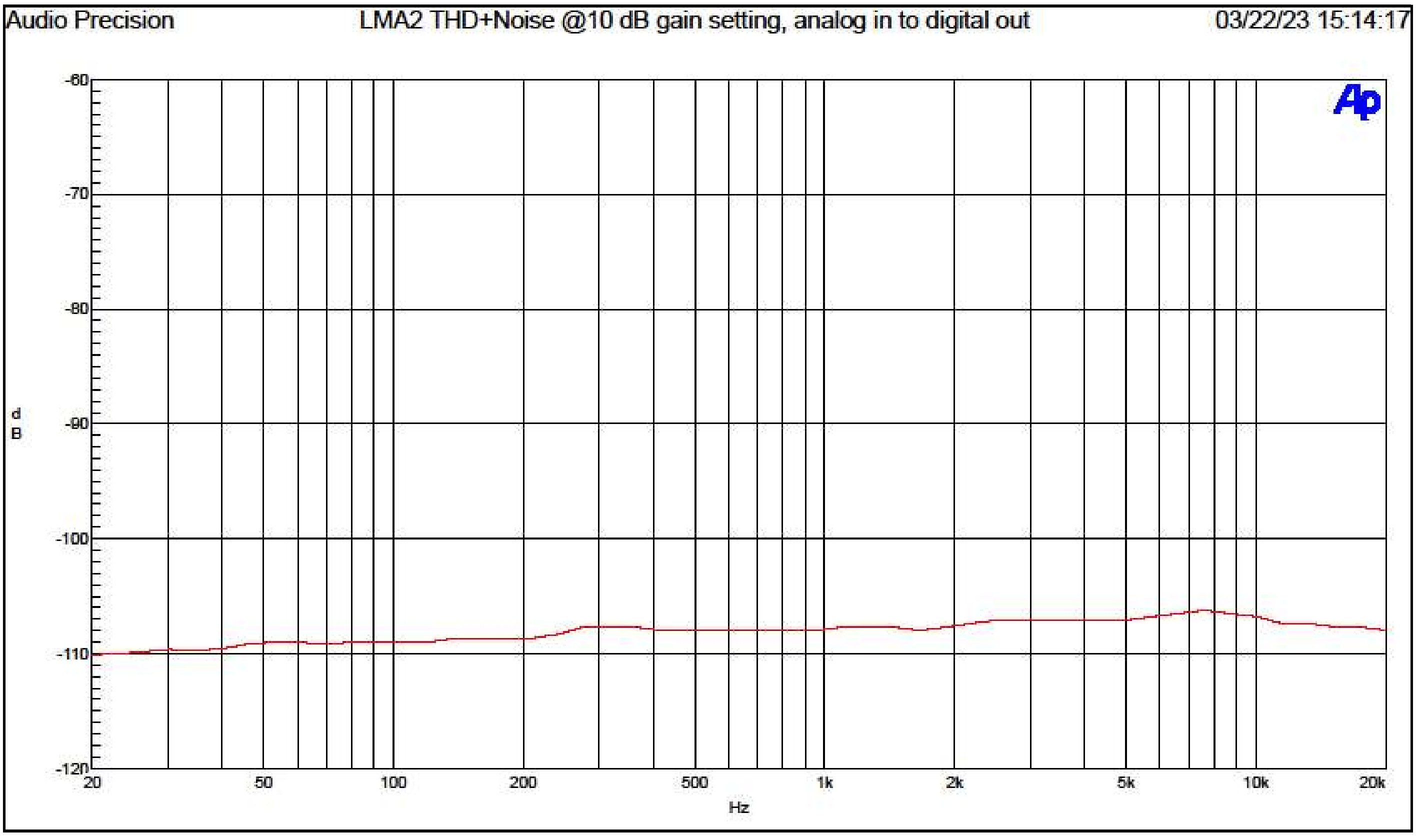
There has always been great focus on distortion, when it comes to judging audio equipment. It can be argued, that any distortion product (2nd, 3rd harmonic etc) below -60 dB is inaudible, simply because the distortion tones are a consequence of the incoming tone, and thus are always masked by it. In some sense this is true, but distortion is more than just about sound, it is also an indicator that shows whether there is something wrong with the circuit or not.

Analog in @ microphone input. Analog out @ analog direct output



THD+Noise versus frequency of the mic/line input circuit. Range 20 Hz-20 kHz, output amplitude +18 dBu
Notice that the distortion is close to linear in the whole frequency range and centered around approx. -110 dB

Analog in @ microphone input. Digital out @ AES output (identical to USB output)



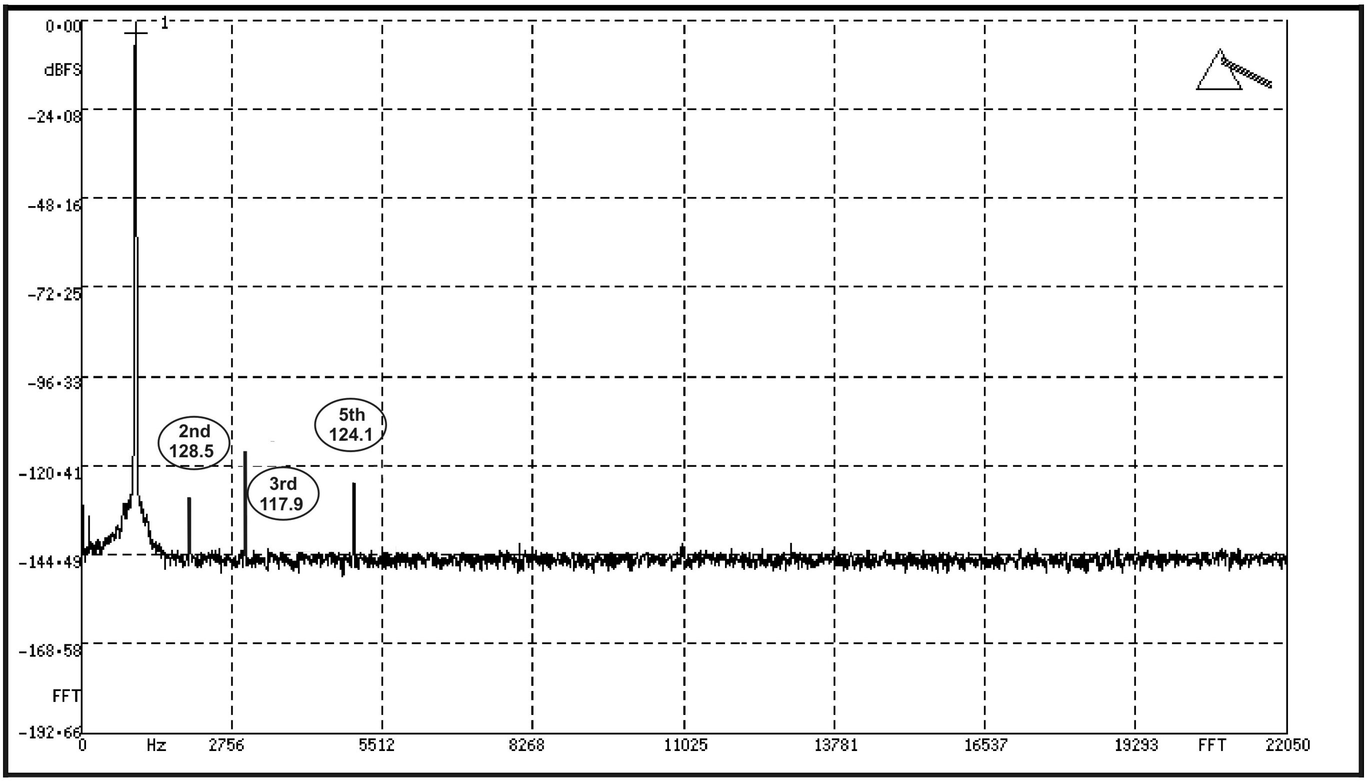
THD+Noise versus frequency. Of the analog circuit + the digital circuit. Range 20 Hz - 20 kHz @ -0.1 dBFS. Only the result @ 44.1 kHz sampling frequency is shown. The distortion characteristics are the same with very little variations at all samplings frequencies.

Distortion in depth analysis. FFT (Fast Fourier Transform)

A more detailed picture of the distortion characteristics can be seen using FFT analysis, which allows the magnitude of the individual distortion components to be quantified. FFT analysis has the advantage of being able to remove random noise in order to reveal the distortion component only.
Noise is a statistical phenomenon, and by reading several samples and adding them together, the FFT analysis is able to reduce the noise to its average value, thus revealing the distortion components otherwise buried in the noise.

In this FFT analysis only the distortion analysis at 44.1 kHz sample rate is included. Distortion characteristics at the other sample rates are identical within less than 0.5 dB.

Analog in @ microphone input. Digital out @ AES output (identical to USB output)



Harmonic distortion measurement, 16 passes of FFT superimposed to suppress noise components.

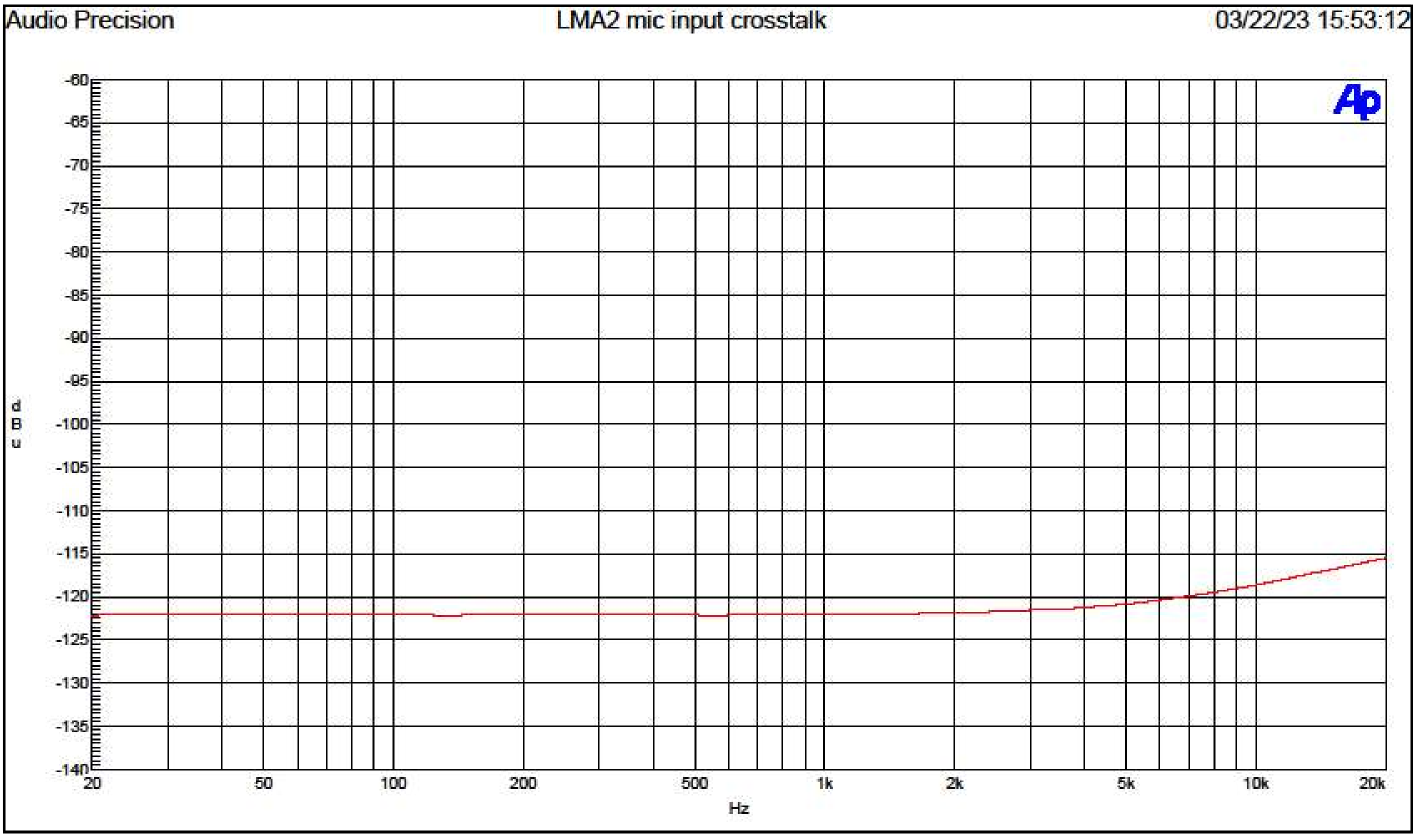
Input is 1 kHz, level is -0.1 dBFS. Sampling rate 44.1 kHz. The resulting distortion sums up to -116.1 dB RMS.

2nd harmonic is 128,5 dB, 3rd harmonic is 117.9 dB, 5th harmonic is 124.1 dB. Only the 2nd, 3rd and 5th harmonic contribute to the distortion RMS result. The harmonics above the 5th are considered negligible.

Crosstalk

Below is the crosstalk measured between adjacent channels (worst case). A sine sweep from 20 - 20 kHz at full amplitude (+18 dBu) was sent to one channel, and the output level was measured at the adjacent channel. The adjacent channel was terminated by 150 ohm at the input in order to eliminate cable transferred crosstalk.

Analog in @ microphone input CH1. Analog out @ analog direct output CH2

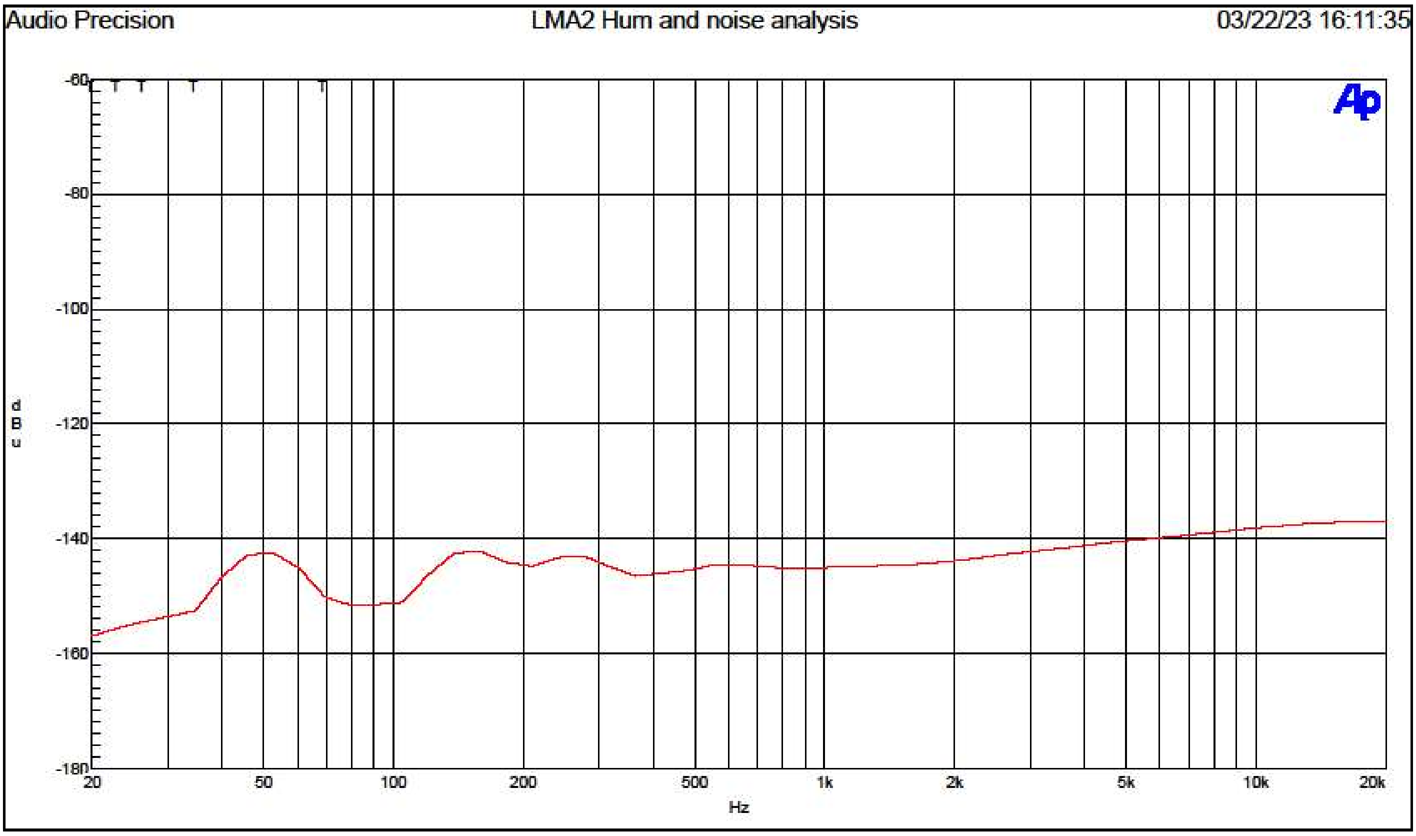


The balanced nature of the LMA2 really pays off. The crosstalk is below -120 dB in the main part of the spectrum. Only above 10 kHz is a slight increase can be seen, reaching approx. -115 dB at 20 kHz

Note: This is a linear and unweighted measurement. Using A-weighting would show a much prettier picture, but that would be cheating!

Hum and noise

Using an analog power supply with an oversized toroidal transformer and linear regulators could be cause for concern. However, not to worry. The carefully designed power circuits create an electromagnetic quiet environment, free from the usual radiation pollution from a switch-mode power supply, and the true balance nature of the analog circuits really pays off. The measurement speaks for itself. There is only negligible power supply noise transferred to the analog outputs.



The graph shows the spectral density of the output noise by measuring with a sweeping 1/3 octave band pass filter across the audio range. The output was measured directly at the output buffers. Digital input was off, and all sampling frequencies were tested.
The noise spectrum is linear and smooth from 1 kHz and up and exhibits the typical characteristics of white noise. Below 1 kHz the 1/f corner noise is dominant. Some small noise components from the power supply can be seen in the critical range from 50 Hz to 400 Hz. The measured result in this range is well below -140 dB, referenced to 0 dBFS.
The total RMS sum of the noise from 20 Hz to 20 KHz at the line output is -118 dB A weighted