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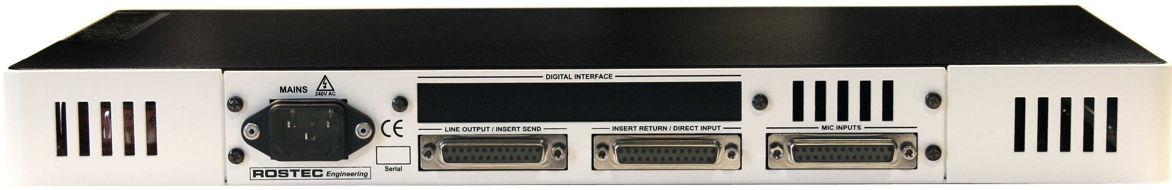
LMA8 Mic/Line Preamplifier



ROSTEC LMA8

8-channel Professional Mic/Line Preamplifier

Revision 5, March 23, 2016

Features

- 8 microphone inputs at the back panel
- 8 high impedance instrument inputs at the front
- 8 balanced line outputs at the back panel
- 8 balanced insert points at the back panel
- Insert points are selectable from the front panel
- Switching done by sealed gold contact relays
- Ultra low noise
- Ultra low distortion
- +30 dBu input headroom
- Automatic switching between instrument input and mic input
- Exceptionally open and transparent sound
- Input circuit has vacuum tube characteristics
- True balanced architecture throughout the unit
- Smooth gain adjustment from +10 dB to +70 dB by potentiometer, no clicks
- Built-in +48 Volts Phantom Power
- Linear low noise power supply. *No Swichmode!!*
- Slot for optional USB or AES digital interface.
- Inputs and outputs are ESD protected to 23 kV, IEC 61000-4-2 and 15 A surge, IEC 61000-4-5.
- Sturdy steel metal casing, electrically and magnetically screened.
- Stand-alone desktop or with mountable 19” rack mount flanges
- Affordable price.

General description

The LMA8 is an 8 channel ultra low noise and ultra low distortion mic/line preamplifier designed with the professional sound engineer in mind. The unusual large input headroom of +30 dBu enables the amp to handle fast transient and large dynamic sound pressure changes with ease. At the same time it reproduces micro-details and environmental depth perspective with a natural openness and impressive accuracy.
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 LMA8 is a purist's dream come true. 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, and it has a transfer characteristic that resembles that of a vacuum tube triode, giving the unit a natural, relaxed and open sound, yet it maintains an extremely fast and totally precise response.

The 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 audio signals at all. This architecture keeps the audio path free from non-linear distortion from currents running in the ground mesh or from “non musical” signals from external electrical fields, power supply noise, crosstalk from other channels etc.

Although the LMA8 is constructed by using modern day's cutting edge technology, the design philosophy is inspired by some of the very best preamps that have been manufactured over the last 50 years.

The basic model is pure analog, aimed at the professional studios that already have high quality AD and DA converters available

Optional USB or AES interface modules are available, using top range digital converters with ultra low distortion and 120 dB dynamic range

Mic inputs and instrument/line inputs

The amp has 8 separate and identical channels. Each channel has two inputs, a microphone input and a high impedance line/instrument input.
The 8 microphone inputs are placed at the back panel and use one 25 pin D-SUB female connector
The 8 instrument/line inputs are placed at the front panel for easy access and use 1/4" standard Jack connectors

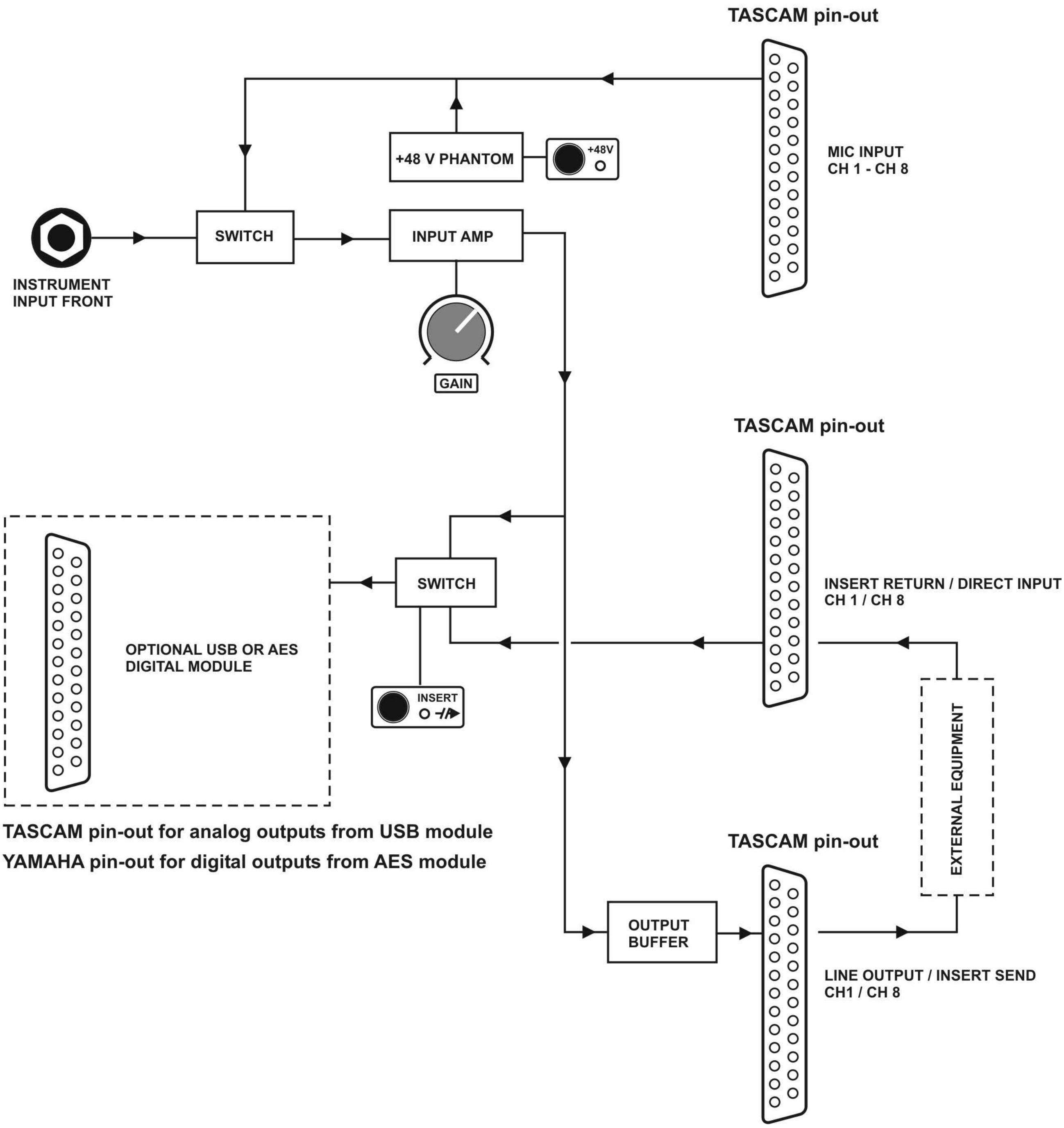
Switching between line input and mic input is automatic. When no Jack is plugged into the front connector, the mic input at the back is active. When a Jack is plugged into the front connector, the unit disables the mic input at the back, activates the line input at the front, and switches to high impedance mode. The high impedance of the line input (1 Mohm/60 pF) is intended for instruments, like guitar or bass, but it is equally well suited for line level equipment, such as keyboards and the like.

Outputs and insert point

Each channel has a buffered balanced line output and a balanced direct input to the digital modules. Each use 25 pin D-SUB female connectors at the back panel.
The 8 line outputs and the 8 direct inputs form the insert points, intended for use with analog equipment. These insert points can be bypassed by pressing the insert point switches at the front panel.

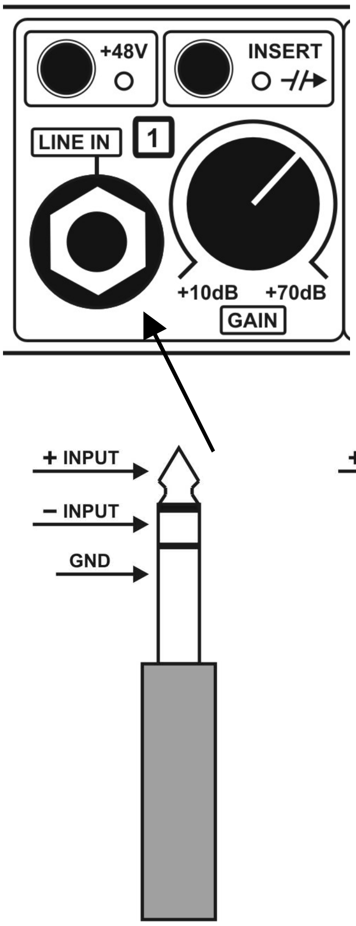
LMA8 SIMPLIFIED SIGNAL FLOW, ONLY ONE CHANNEL SHOWN

Pin number connections are just illustrative



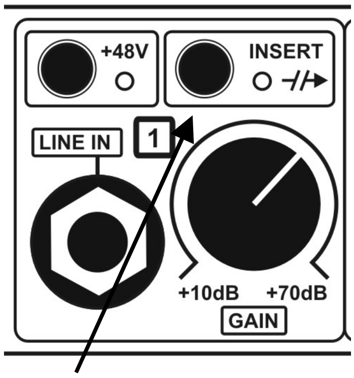
TASCAM pin-out for analog outputs from USB module
YAMAHA pin-out for digital outputs from AES module

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 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 page 8, *Remote ground connection*



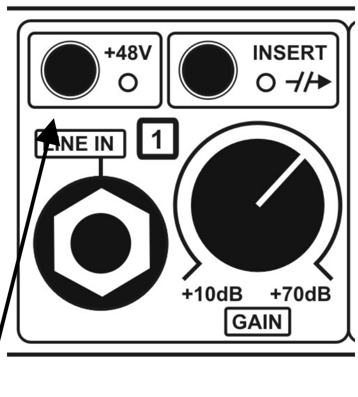
Insert point switch

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

When the LED is ON, the signal chain between the preamp and the digital module is “open” which means that external equipment can then be inserted between line output/insert send and the insert return/direct input.

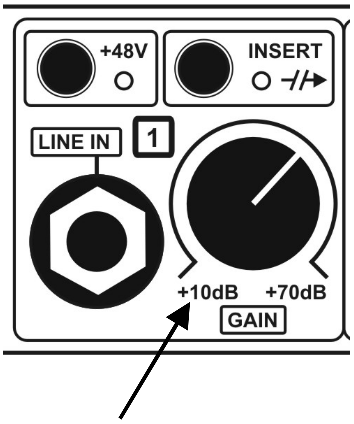
When the LED is OFF, the signal is routed directly from the preamp to the digital module. The insert return/direct input is then inactive.
The line output/insert send is not affected, and is always active

When no digital module is installed, the switch has no function. Pressing the switch will momentarily turn the LED on, but the LED will automatically turn off again after 2 seconds.



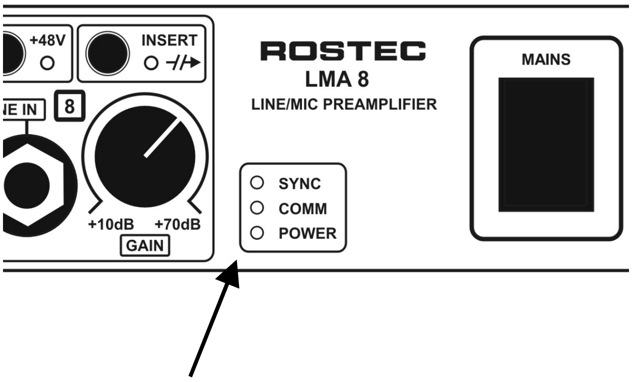
+48V Phantom Voltage

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



__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 both instrument/line input and microphone input.
The gain range is +10 to +70 dB from input to the line output on the D-SUB connector at the back.

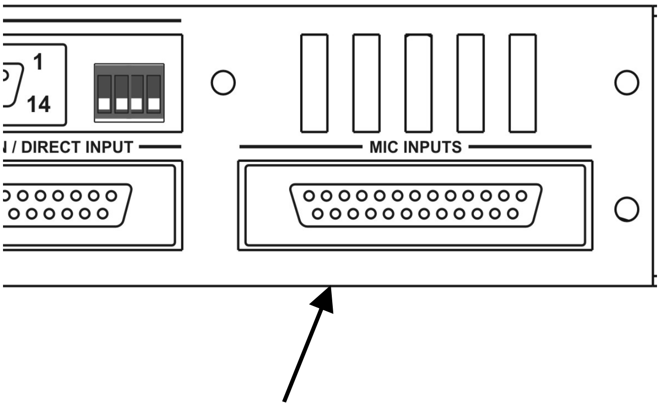


Status indicators

These LEDs are status indicators for the digital interface module (USB or AES) and for Power Good.

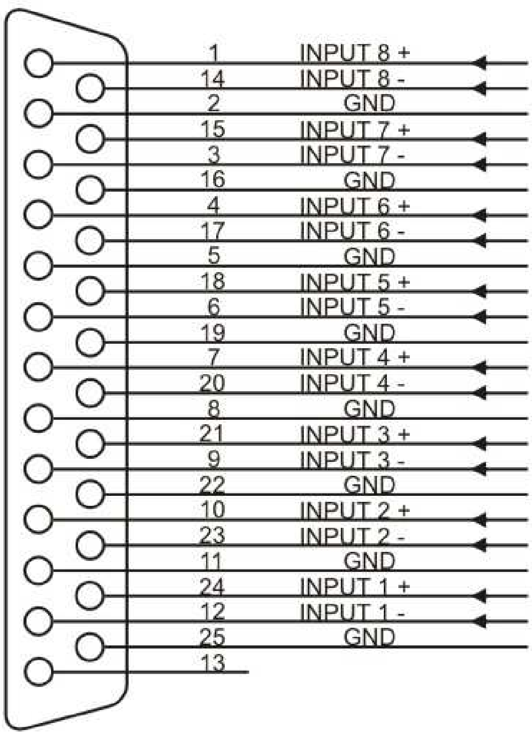
SYNC indicates a solid lock to the incoming word clock.
COMM indicates a good communication link to the DAW
POWER indicates that power is on

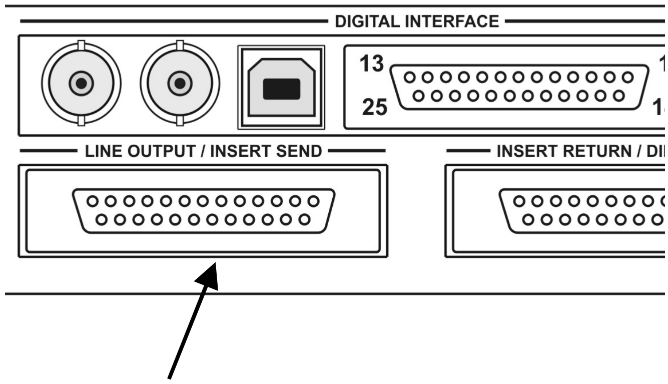
The indicators should be steady on during normal operation. When no digital module is installed, only the POWER LED is active.



Analog inputs (mic inputs)

The analog input D-SUB connector is for the microphone inputs. The maximum input is +30dBu. The input impedance is 6 kohm
The 25 pin D-SUB connector follows the TASCAM standard as shown below.
Observe that channel 8 is at the beginning of the pin numbers and channel 1 is at the end.
It seems a little awkward, but it is the reigning industrial standard. Following this standard makes it easier to interface with off-the-shelf cables to other multi channel equipment, like AD/ DA converters, mixing consoles etc



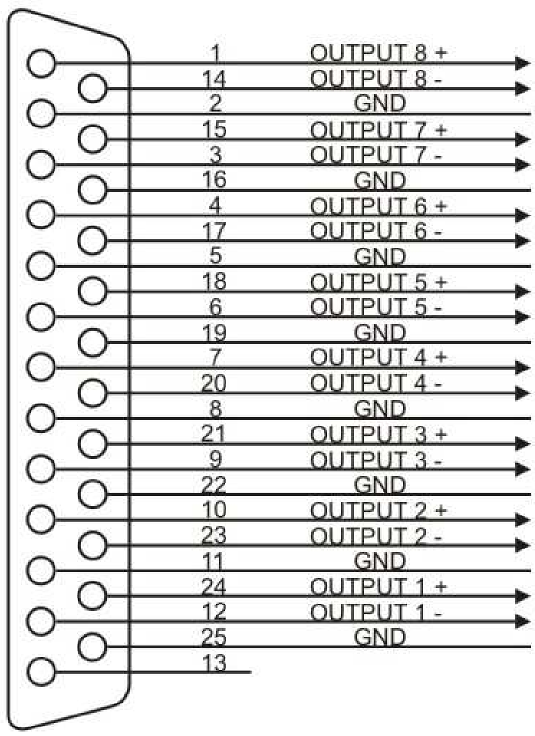


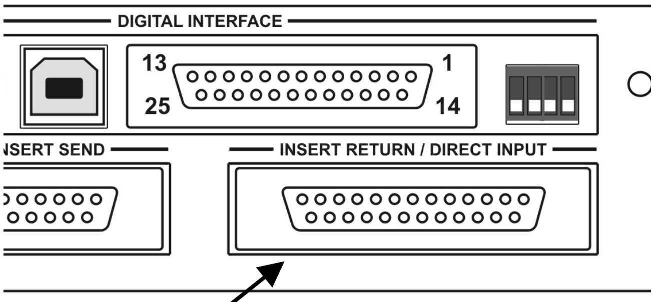
Analog outputs (line outputs)

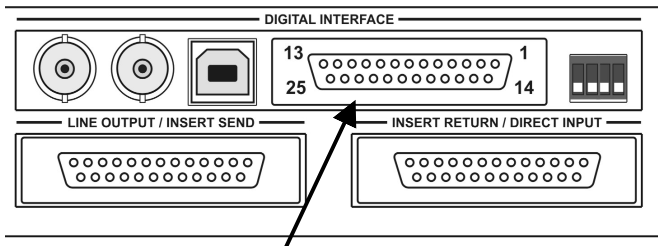
The analog output D-SUB connector is for the balanced line outputs. The output buffers are of industrial grade transformerless balanced floating configuration. For unbalanced operation, the negative output must be shorted to ground

Maximum output is +30 dBu in balanced mode.
Maximum output is +24 dBu in unbalanced mode.

The 25 pin D-SUB connector follows the TASCAM standard.







Direct input

The analog input D-SUB connector is for the balanced direct line inputs to any installed digital module.
With no digital module installed, the connector is inactive
The inputs are electronically balanced. For unbalanced operation, the negative input should be shorted to ground.

The input impedance is 10 kohm
Maximum input is +30 dBu in balanced mode.
Maximum input is +24 dBu in unbalanced mode.

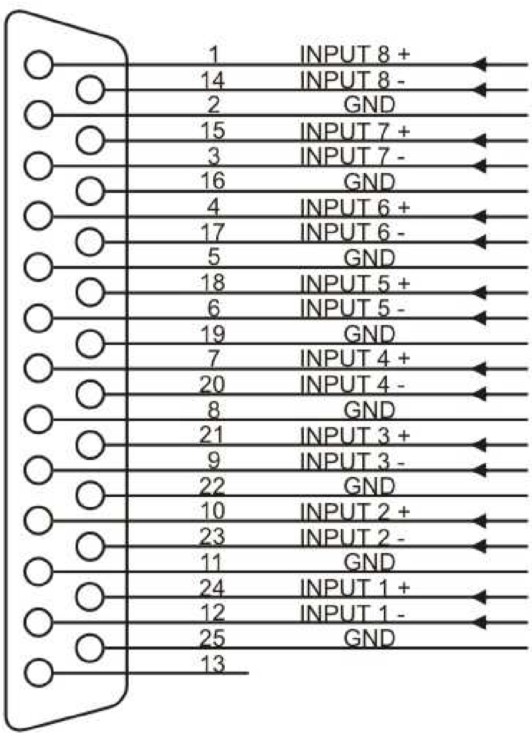
The 25 pin D-SUB connector follows the TASCAM standard.

Digital interface slot

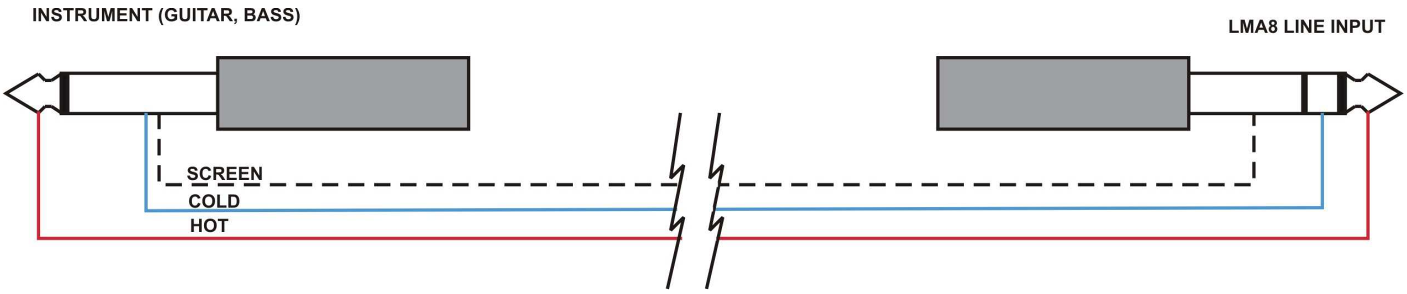
This is the slot for the optional digital interface module, shown here with the USB module installed

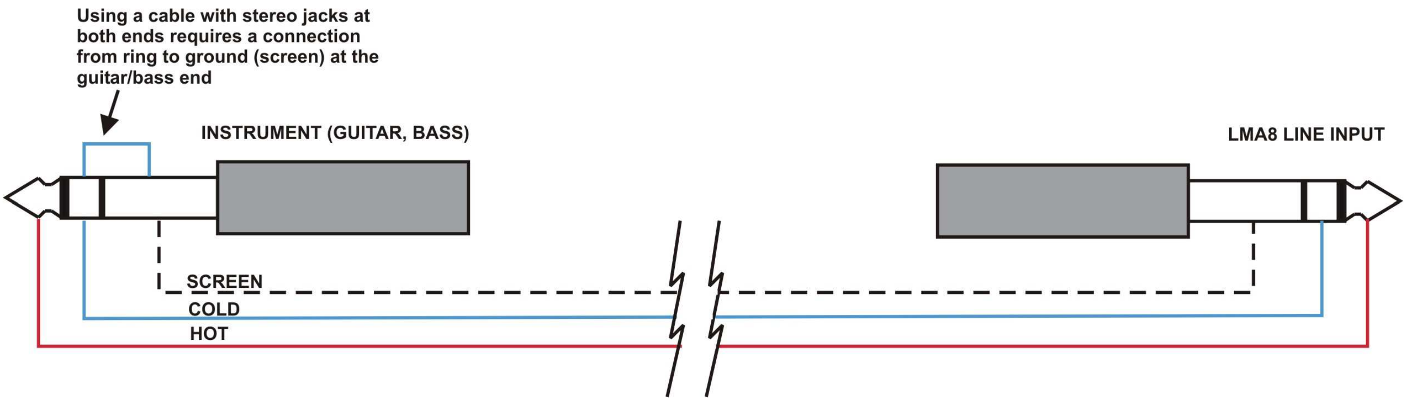
The interface module is installed by opening op the box and fastening the module with screws and connecting it with ribbon cables.

The procedure is simple and straight forward for a qualified technician. It is described in the technical manuals for the USB and AES modules.



Remote ground connection





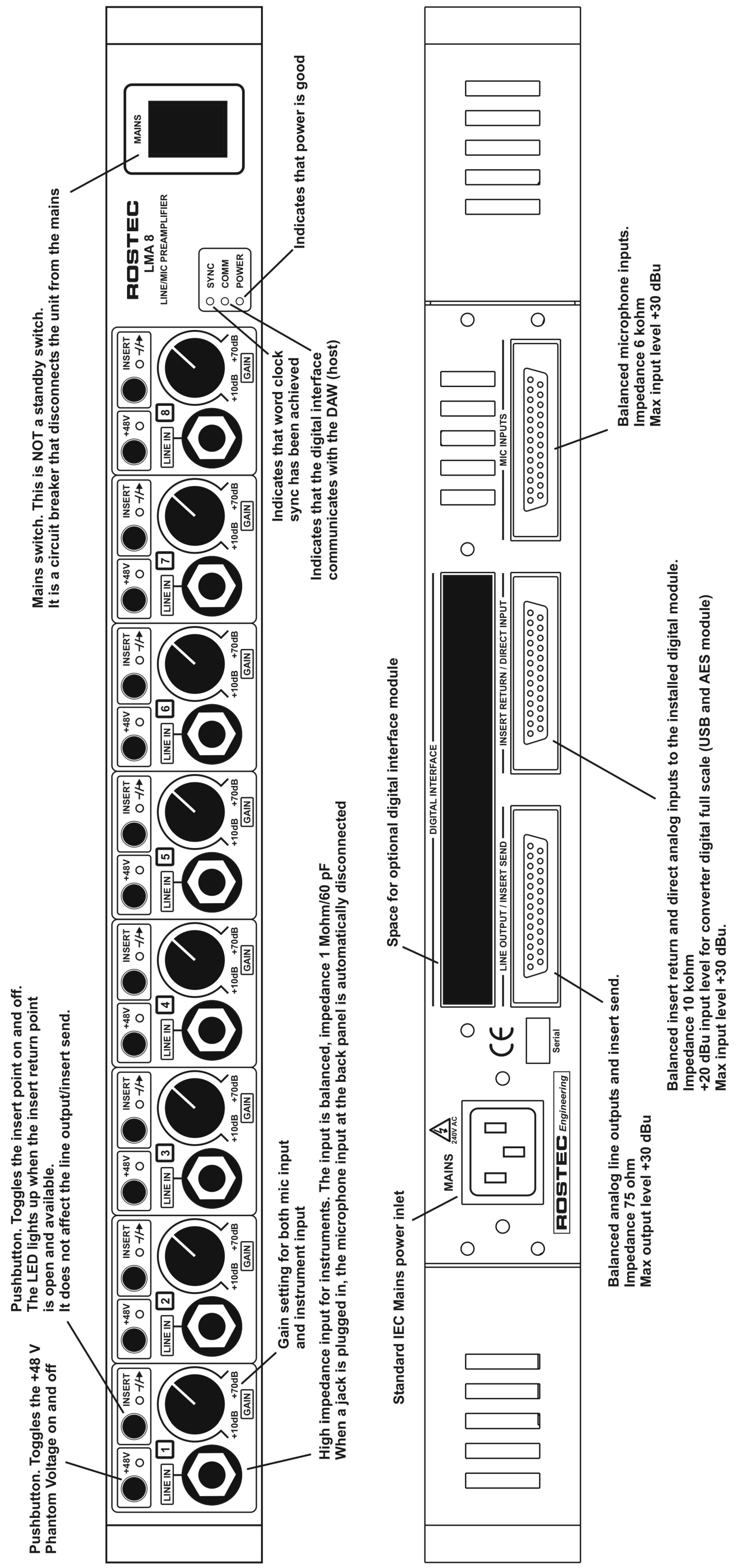
The balanced instrument/line input of the LMA8 opens up the unique possibility to use a pseudo balanced configuration with instruments like guitar, bass etc. The configuration will cancel out a large portion of the electrical noise induced in the cables. This is quite advantageous when using long cables 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 LMA8 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.

Front and back panel quick guide



Mechanical and electrical specifications

Dimensions with rack mounts: Width 19 inch, height 1U (44 mm), depth 210 mm
Dimension as desktop: Width 420 mm, height 42 mm, depth 210 mm
Weight: 5.0 kg
Power: 180 - 253 VAC, 50-60 Hz, 15 Watts
ESD: Protected to 23 kV, IEC 61000-4-2 and 15 A surge, IEC 61000-4-5

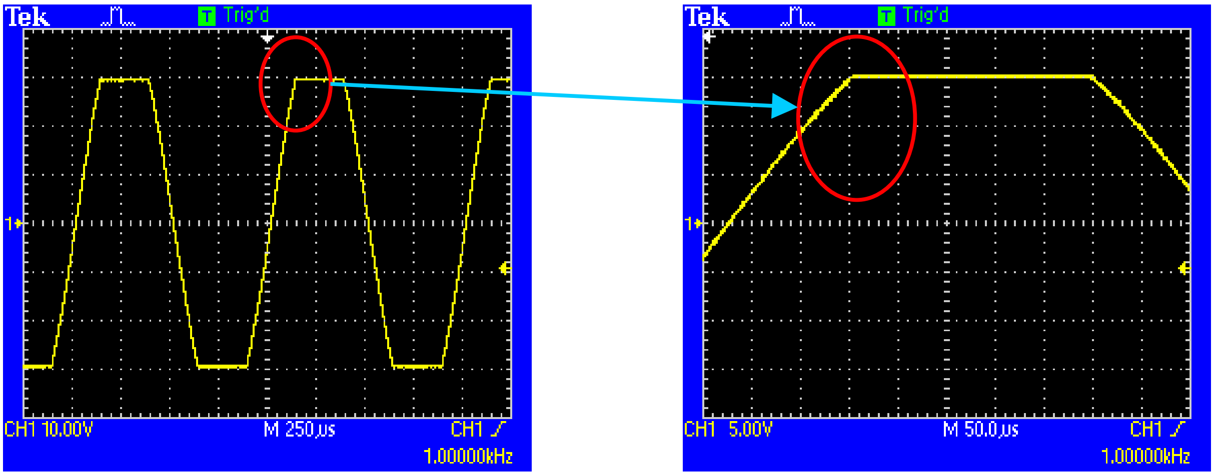
Mic input impedance: 6 kohm
Line input impedance: 1 Mohm/60 pF
Max input level balanced, mic or line: +30 dBu
Max input level unbalanced, line: +24 dBu
Max input level balanced, direct input: +30 dBu
Max input level unbalanced, direct input: +24 dBu
Max output level, balanced output: +30 dBu
Max output level, unbalanced output +24 dBu
Gain, input to balanced output: +10 dB to +70 dB
Input noise: -134 dBu (A weighted, 22 Hz - 22 kHz)
Input noise: -131 dBu unweighted
Frequency response: 5 Hz - 200 kHz, +/- 0.1 dB
Crosstalk: -120 dB, 20 Hz - 20 kHz, input terminated by 150 ohm

<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

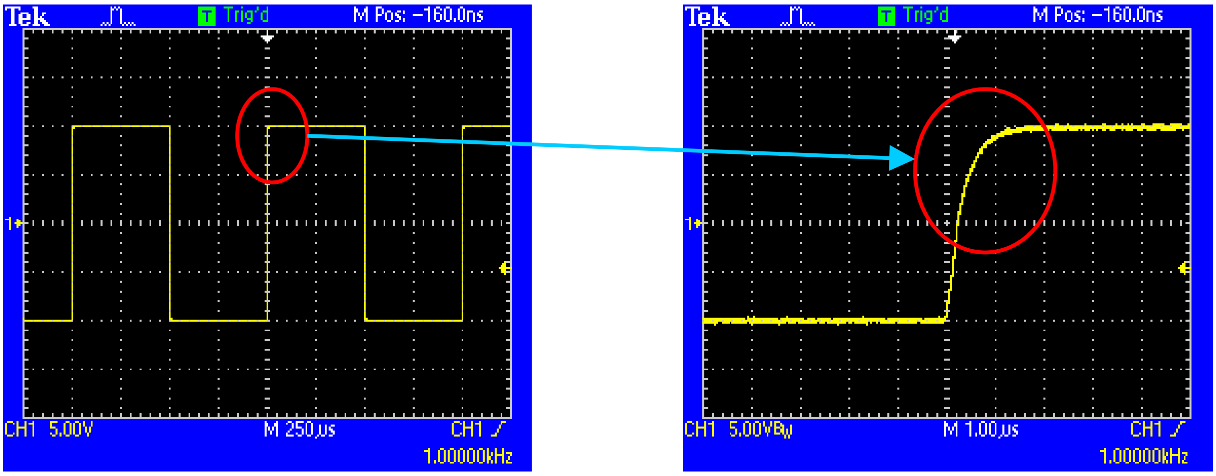
Technical section

Clipping characteristics



The LMA8 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 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 LMA8. When subjected to at steep transient (in this case a step impulse 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 +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.

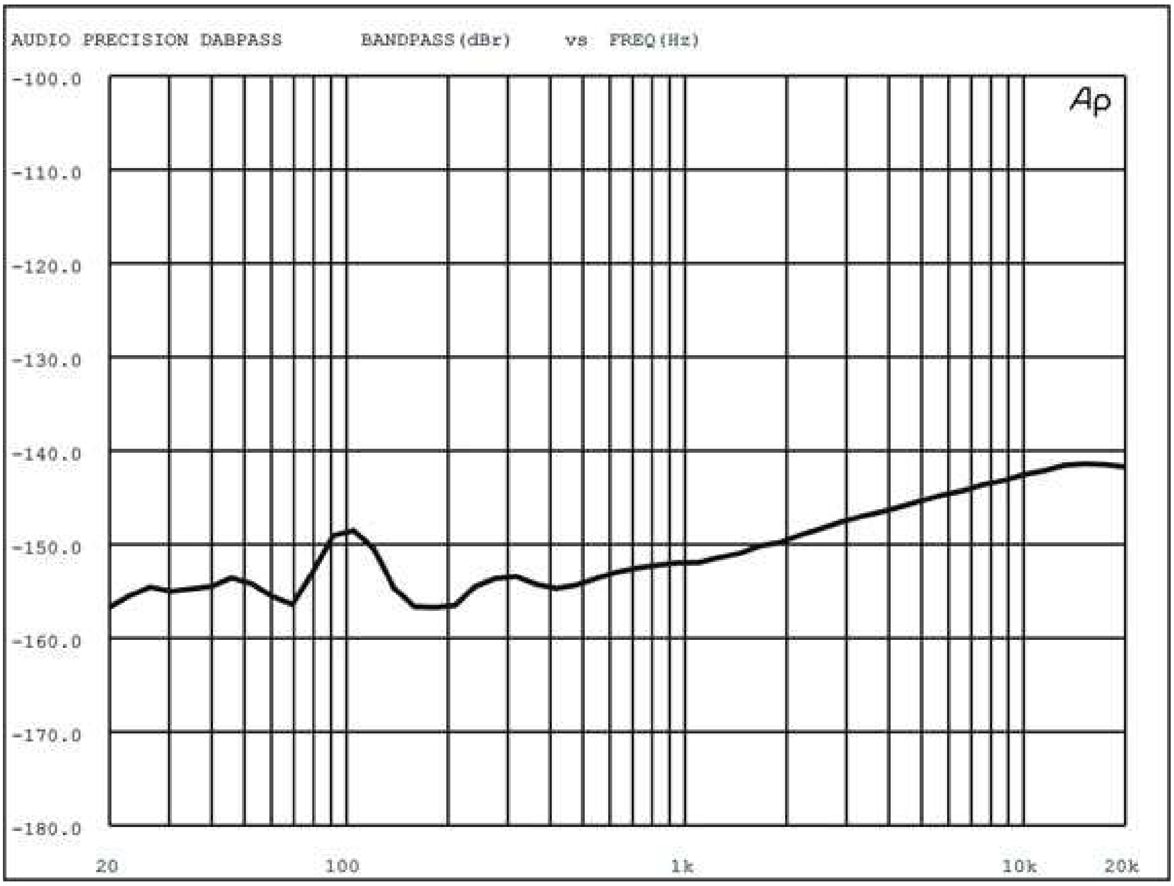
But not to worry; LMA8 has a mechanism in place to prevent such unwanted HF feedback.

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 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

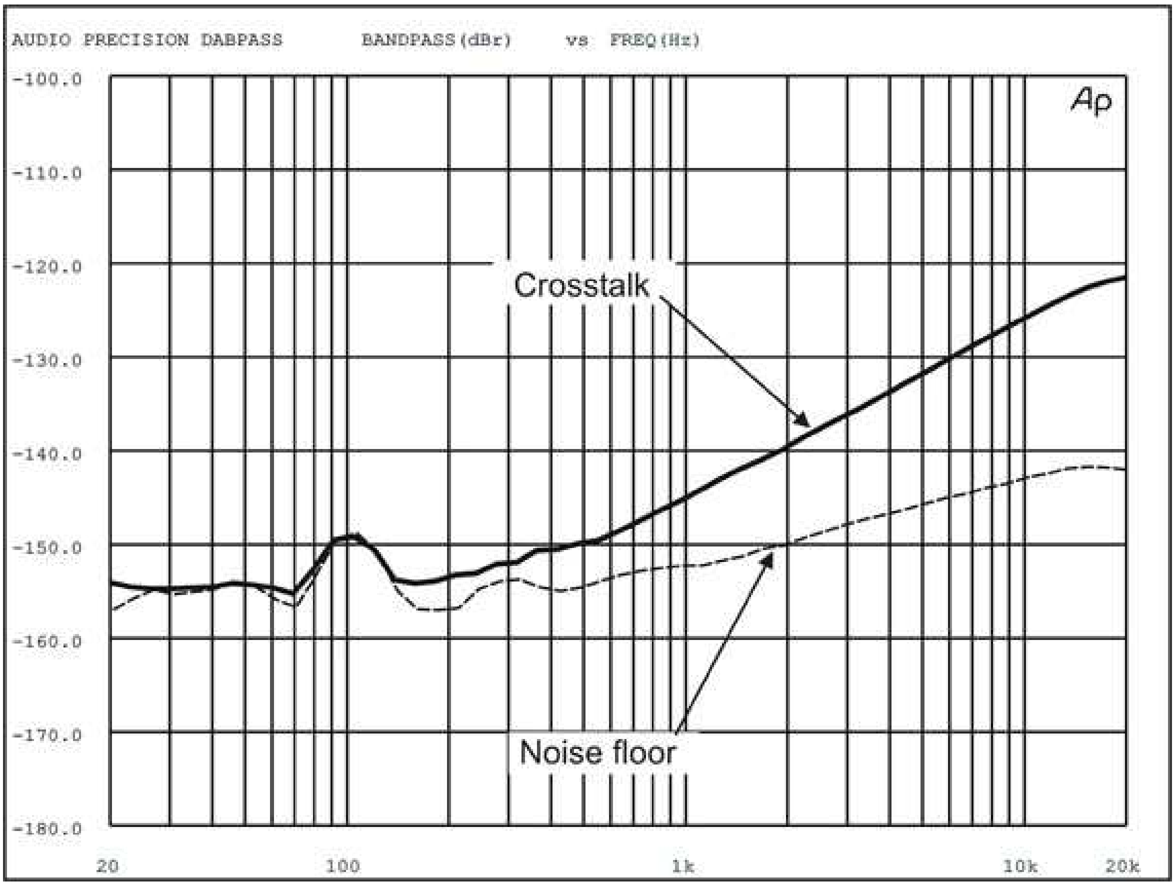
The change of the frequency response is not audible and it does not affect the internal gain structure, noise figure or distortion figure.

Noise performance and crosstalk



Input noise

The graph shows the spectral density of the input noise by measuring with a sweeping 1/3 octave bandpass filter across the audio range.
The input was terminated by 150 ohms and the gain was set to max (+70 dB)
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. A noise component from the power supply can be seen at 100 Hz at a level of approx. -148 dBu
The rms sum of the noise from 22 - 22 kHz is -134 dBu A weighted, or -131 dBu unweighted.



Crosstalk

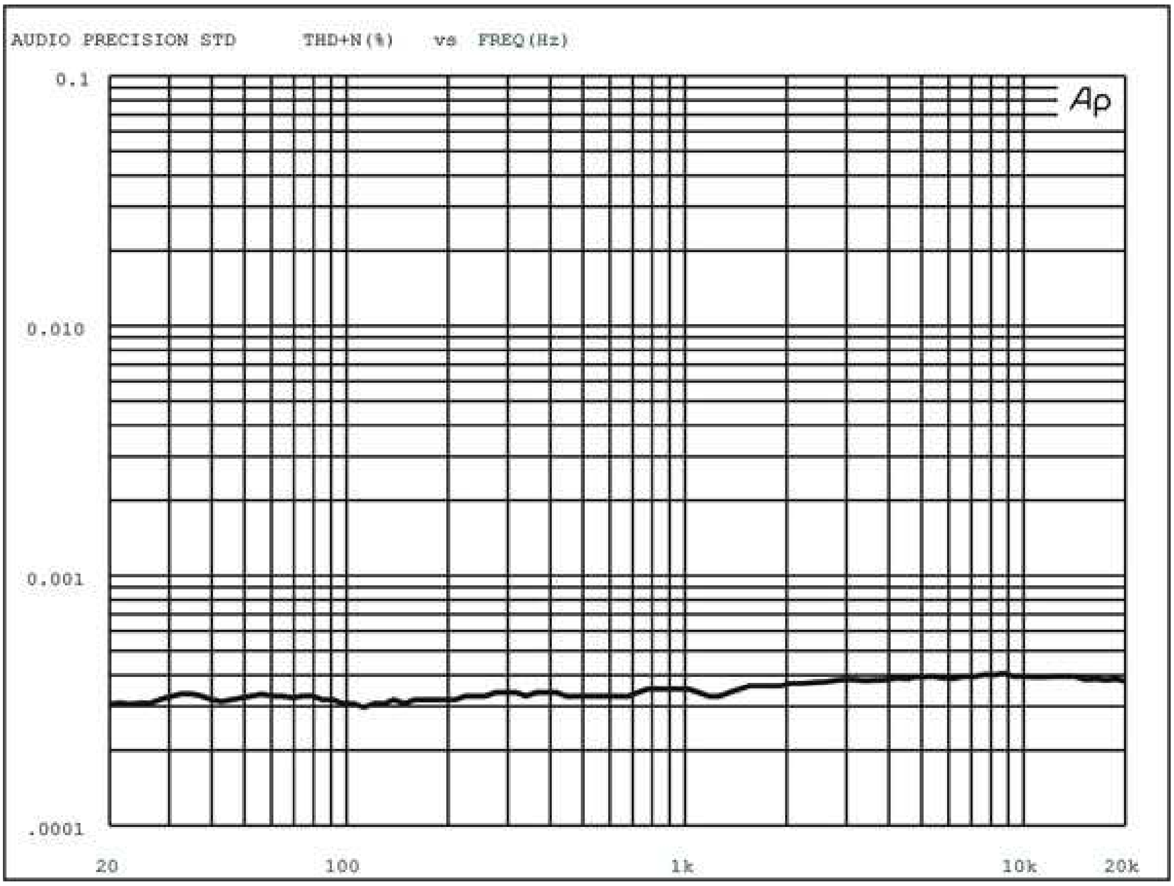
The graph shows crosstalk between adjacent channels. A +16 dBu signal was sent to channel 1, gain setting +10 dB. The output level of channel 1 was +26 dBu.
The input of channel 2 was terminated by 150 ohms, and the signal level was measured at the output of channel 2.

The graph shows virtually no crosstalk below 100 Hz, which indicates that there is no transfer of audio signals due to signal dependent ground currents
The crosstalk increases moderately at higher frequencies, but it is still below -120 dBu at 20 kHz. This indicates a small capacitive audio signal transfer internally between components.

Channel 1 is blasting at +26 dBu and channel 2 sees only -120 dBu at the input. This clearly demonstrates the benefits of a <u>*true balanced architecture.*</u>

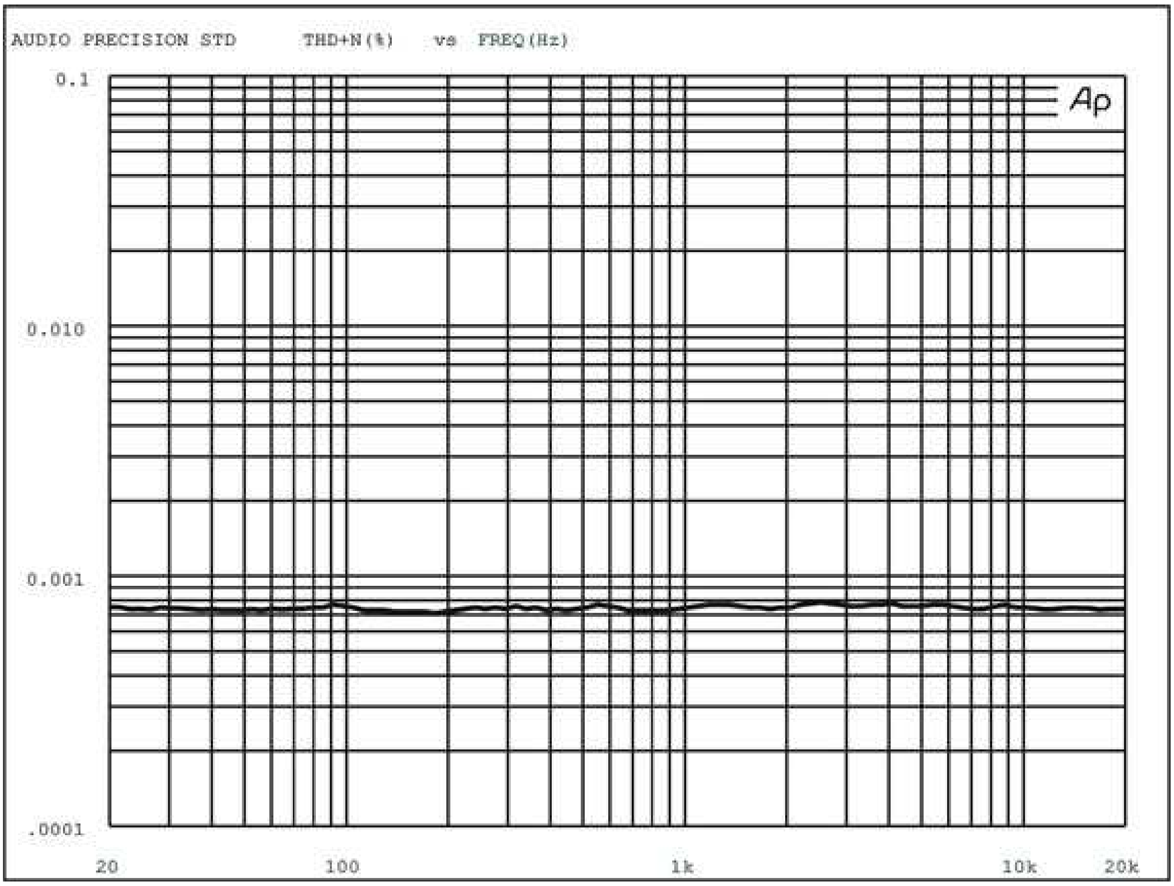
Classical Distortion Analysis

The standard analysis of distortion in audio circuits has traditionally been "Total Harmonic Distortion + Noise".
This method has only limited usefulness in this case, because the distortion of the LMA8 is so low, that it is masked by noise at higher gain settings.
The LMA8's input noise is among the lowest today's technology can offer, and the distortion of the unique input amplifier circuit is so low, that it can only be quantified by classical analysis at lower gain settings. At higher gain settings, a FFT analysis is required.



Distortion + noise at +10 dB gain

The THD+N is below 0.0004 % in the whole audio range. The measured THD+N result shows good linearity from 20 Hz to 20 kHz; only a slight increase above 2 kHz is seen.
The input noise contribution at +10 dB gain corresponds to a level of approx. 0.00007 %.
Thus it can be seen that the graph **mainly** represents the distortion of the circuit.



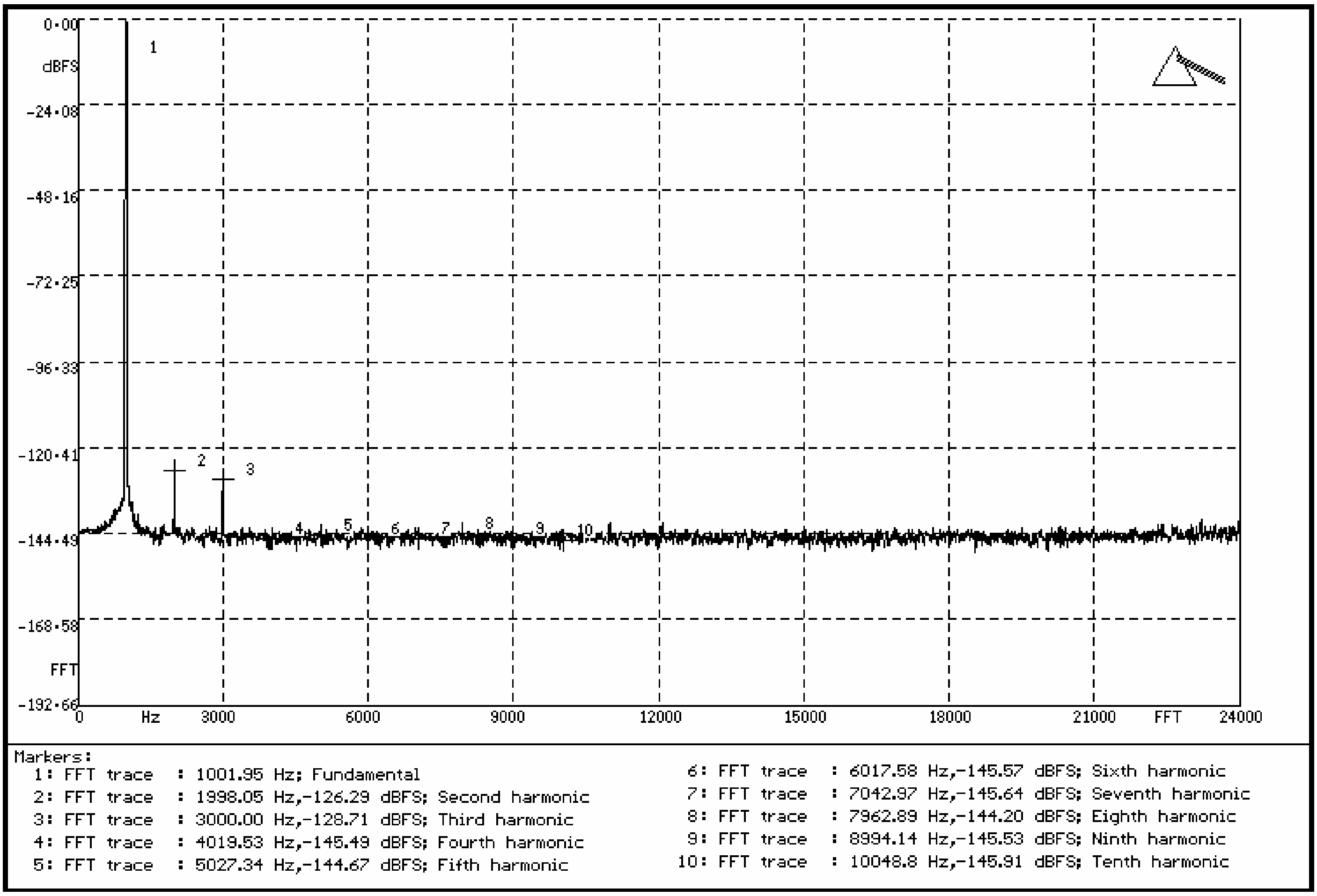
Distortion + noise at +30 dB gain

The THD+N is below 0.0008 % in the whole audio range. The measured THD+N result is very close to linear from 20 Hz - 20 kHz
The input noise contribution at +30 dB gain corresponds to a level of approx. 0.0007 %.
The distortion of the circuit is still below 0.0004 %.
Thus it can be seen that the graph **mainly** represents the noise of the circuit.
Increasing the gain setting further has little effect on the distortion figure, but will only add noise to the measurement.

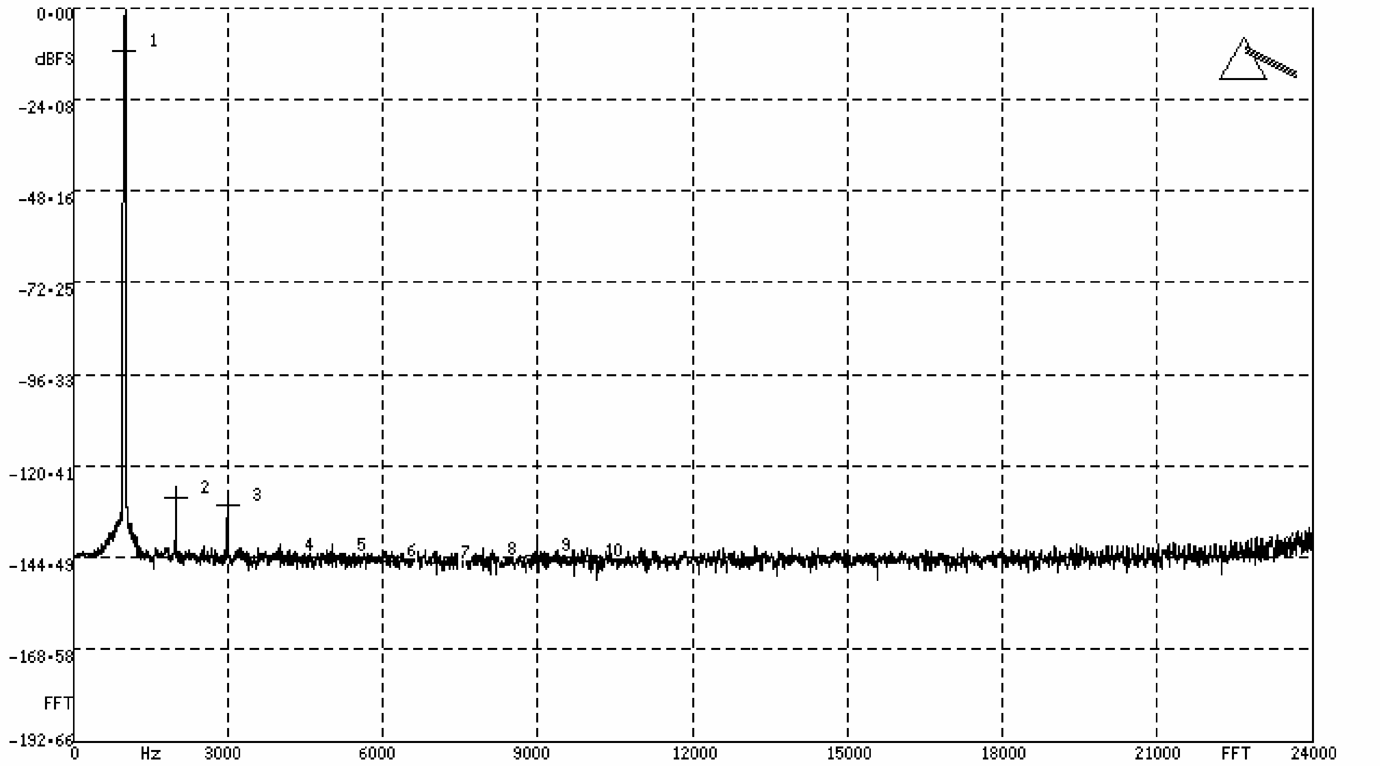
FFT distortion analysis

FFT analysis has the ability to detect distortion that is buried in the noise. Noise is a statistical phenomenon, and by reading several samples and adding them together, the FFT analysis is able reduce the noise to its average value, thus revealing the distortion components otherwise buried in the noise.

Below are plots of the distortion characteristics at various gain settings. Distortion levels are listed for 2nd to 10th harmonic and are quantified in dBFs. As a guideline, 120 dBFs equals 0.0001 %, 140 dBFs equals 0.00001 %

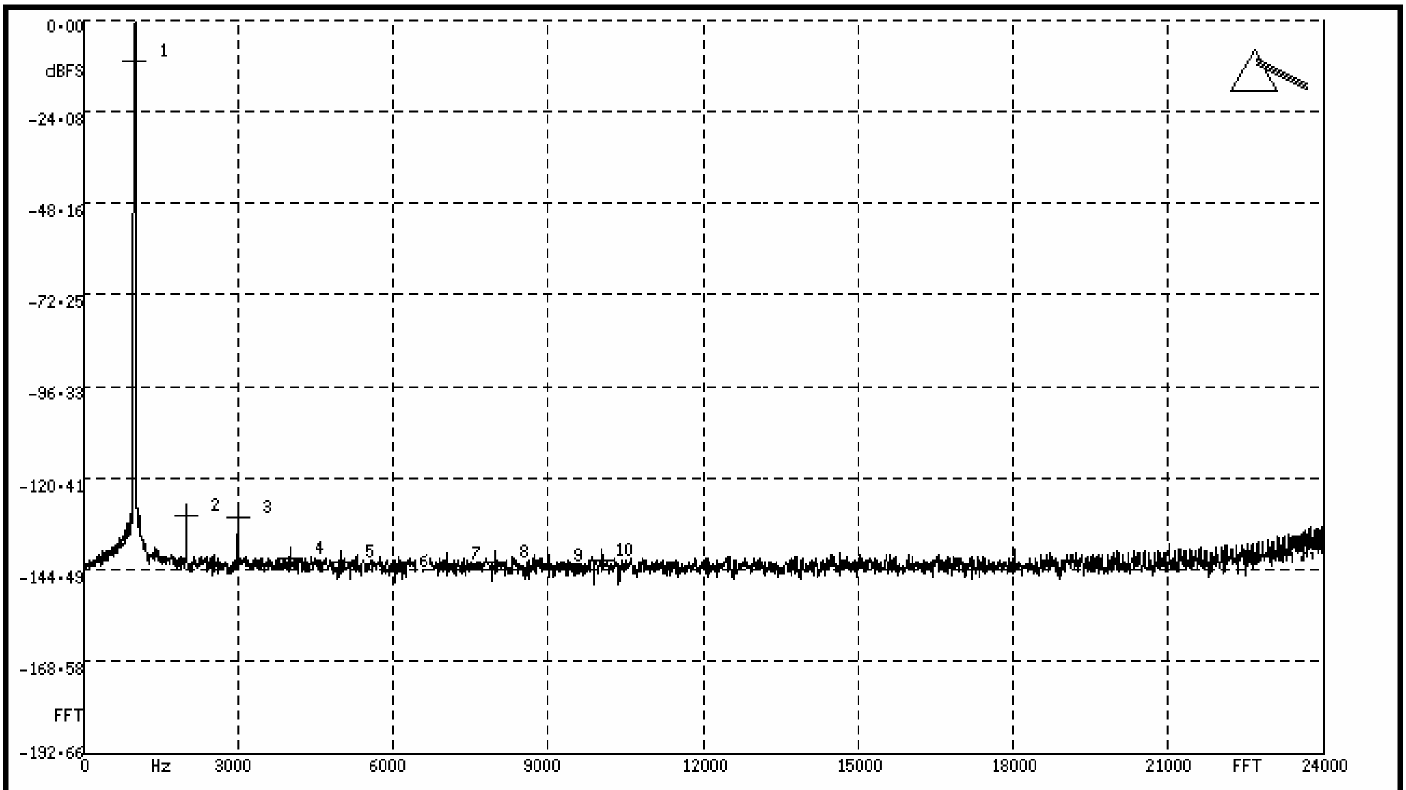


Gain +10 dB, frequency 1 kHz. The rms sum of all distortion components (THD) is 0.00013 %



Markers:
1: FFT trace : 1001.95 Hz; Fundamental
2: FFT trace : 1998.05 Hz,-128.33 dBFS; Second harmonic
3: FFT trace : 3000.00 Hz,-130.41 dBFS; Third harmonic
4: FFT trace : 4019.53 Hz,-143.73 dBFS; Fourth harmonic
5: FFT trace : 5027.34 Hz,-143.68 dBFS; Fifth harmonic
6: FFT trace : 6017.58 Hz,-144.85 dBFS; Sixth harmonic
7: FFT trace : 7042.97 Hz,-145.36 dBFS; Seventh harmonic
8: FFT trace : 7962.89 Hz,-144.70 dBFS; Eighth harmonic
9: FFT trace : 8994.14 Hz,-143.39 dBFS; Ninth harmonic
10: FFT trace : 10048.8 Hz,-145.00 dBFS; Tenth harmonic

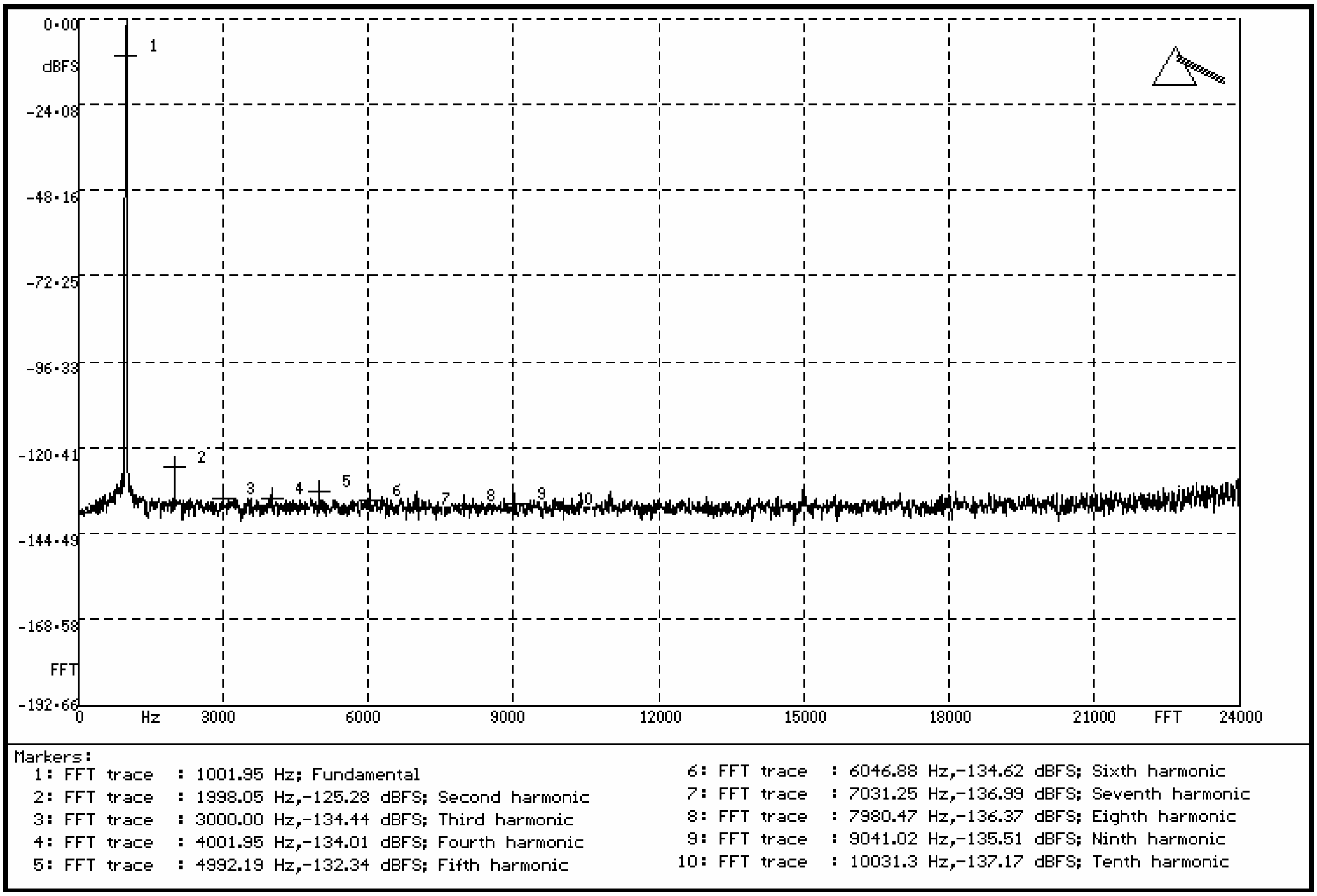
Gain +20 dB, frequency 1 kHz. The rms sum of all distortion components (THD) is 0.00014 %



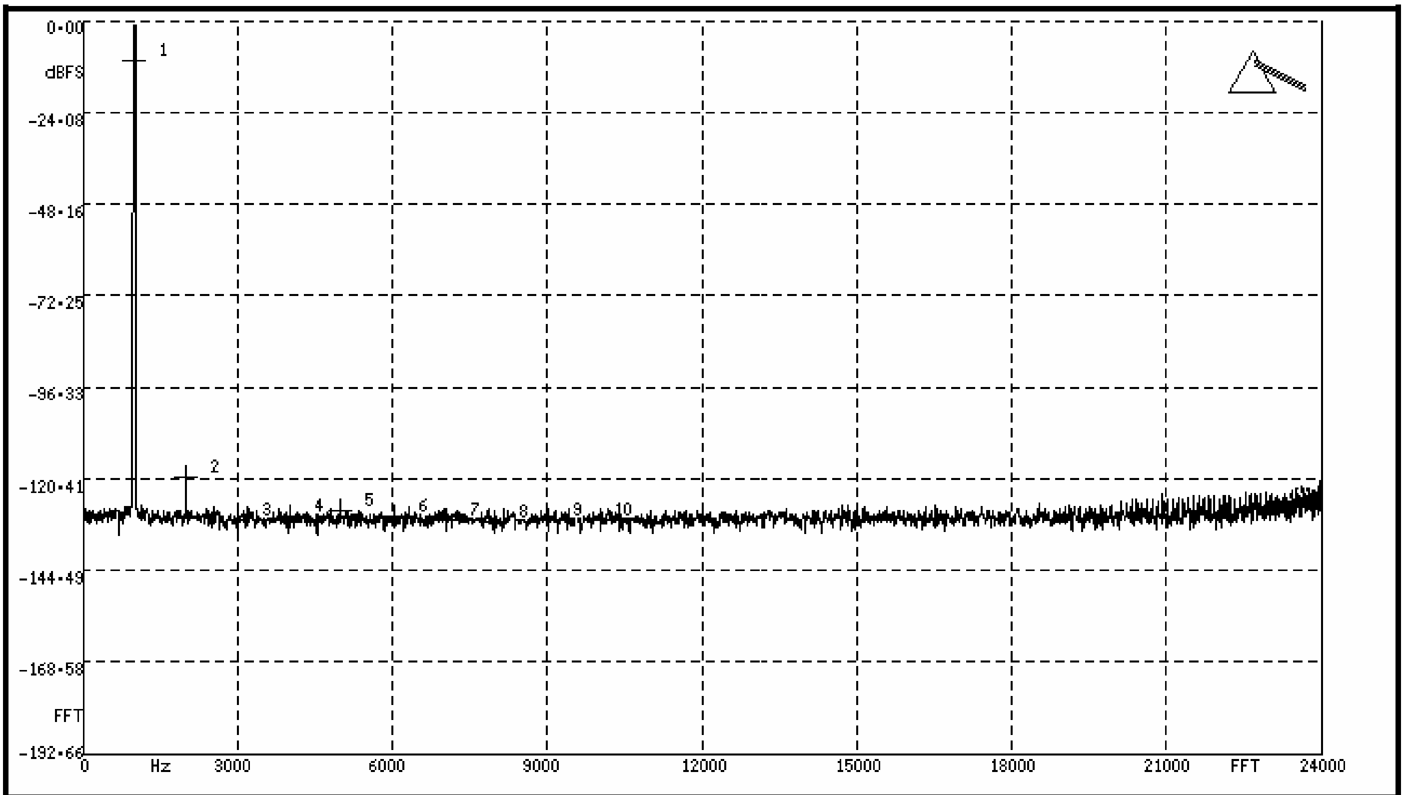
Markers:

1: FFT trace : 1001.95 Hz; Fundamental
2: FFT trace : 1998.05 Hz,-129.63 dBFS; Second harmonic
3: FFT trace : 3000.00 Hz,-130.54 dBFS; Third harmonic
4: FFT trace : 4001.95 Hz,-141.07 dBFS; Fourth harmonic
5: FFT trace : 4992.19 Hz,-142.08 dBFS; Fifth harmonic
6: FFT trace : 6046.88 Hz,-144.35 dBFS; Sixth harmonic
7: FFT trace : 7031.25 Hz,-142.60 dBFS; Seventh harmonic
8: FFT trace : 7980.47 Hz,-142.14 dBFS; Eighth harmonic
9: FFT trace : 9041.02 Hz,-142.94 dBFS; Ninth harmonic
10: FFT trace : 10031.3 Hz,-141.79 dBFS; Tenth harmonic

Gain +30 dB, frequency 1 kHz. The rms sum of all distortion components (THD) is 0.00015 %



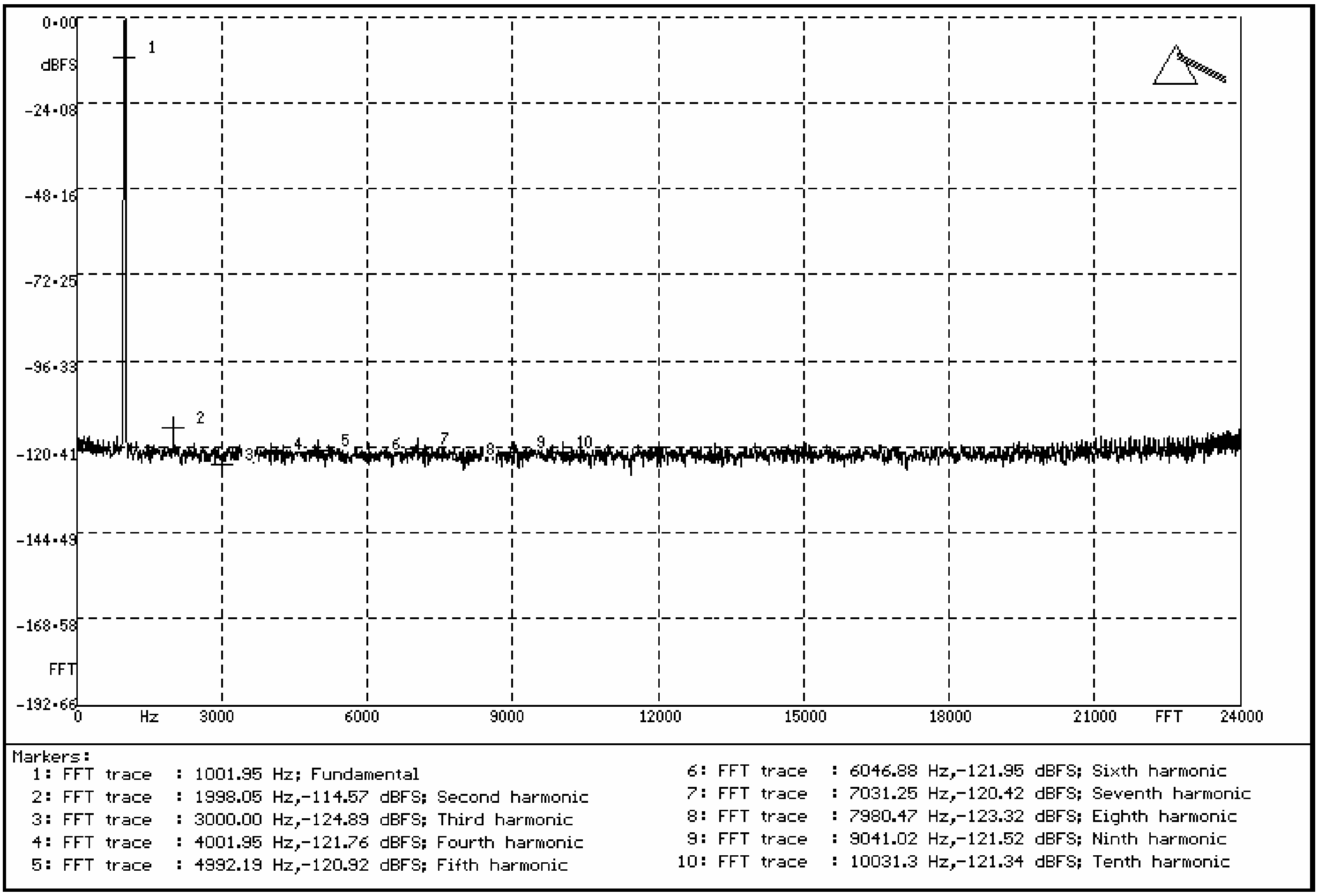
Gain +40 dB, frequency 1 kHz. The rms sum of all distortion components (THD) is 0.00025 %



Markers:

1: FFT trace : 1001.95 Hz; Fundamental
2: FFT trace : 1998.05 Hz,-119.46 dBFS; Second harmonic
3: FFT trace : 3000.00 Hz,-130.61 dBFS; Third harmonic
4: FFT trace : 4001.95 Hz,-130.02 dBFS; Fourth harmonic
5: FFT trace : 4992.19 Hz,-128.20 dBFS; Fifth harmonic
6: FFT trace : 6046.88 Hz,-129.68 dBFS; Sixth harmonic
7: FFT trace : 7031.25 Hz,-130.80 dBFS; Seventh harmonic
8: FFT trace : 7980.47 Hz,-131.21 dBFS; Eighth harmonic
9: FFT trace : 9041.02 Hz,-130.84 dBFS; Ninth harmonic
10: FFT trace : 10031.3 Hz,-130.86 dBFS; Tenth harmonic

Gain +50 dB, frequency 1 kHz. The rms sum of all distortion components (THD) is 0.00045 %



Gain +60 dB, frequency 1 kHz. The rms sum of all distortion components (THD) is 0.00118 %