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LMA8 Mic/Line Preamplifier



# ROSTEC LMA8

## 8-channel Professional Mic/Line Preamplifier

## Revision 4, March 25, 2013

**General description**



**Operational description**





**Technical section**

***Features***

- 8 microphone inputs on the back.
- 8 high impedance instrument inputs on the front
- 8 balanced high level line outputs
- 2 unbalanced instrument outputs
- Ultra low noise
- Ultra low distortion
- +30 dBu input and output headroom
- Automatic switching between instrument input and mic input
- Exceptional open and transparent sound
- Input circuit has vacuum tube characteristics
- True balanced architecture throughout
- Gain adjustment by potentiometer, no clicks
- Selectable mic input impedance
- Built-in +48 Volts Phantom Power
- Linear low noise power supply
- 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.
- 19" sturdy steel metal casing.
- 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 no input clipping can be experienced. The input circuit has a +30 dBu headroom, and because there is always 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 the architecture is fully balanced throughout the unit, making the amp immune to outside electrical disturbances, power supply noise, crosstalk from adjacent channels etc. The input circuit has a transfer characteristics that resembles that of a vacuum tube triode, giving the unit a natural, relaxed and open sound, yet it is extremely fast and totally precise.

Although the LMA8 is constructed of 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 studio that already has invested in high quality AD and DA converters.
A USB or an AES interface is available, featuring top range digital converters with 120 dB dynamic range.

***Inputs and outputs***

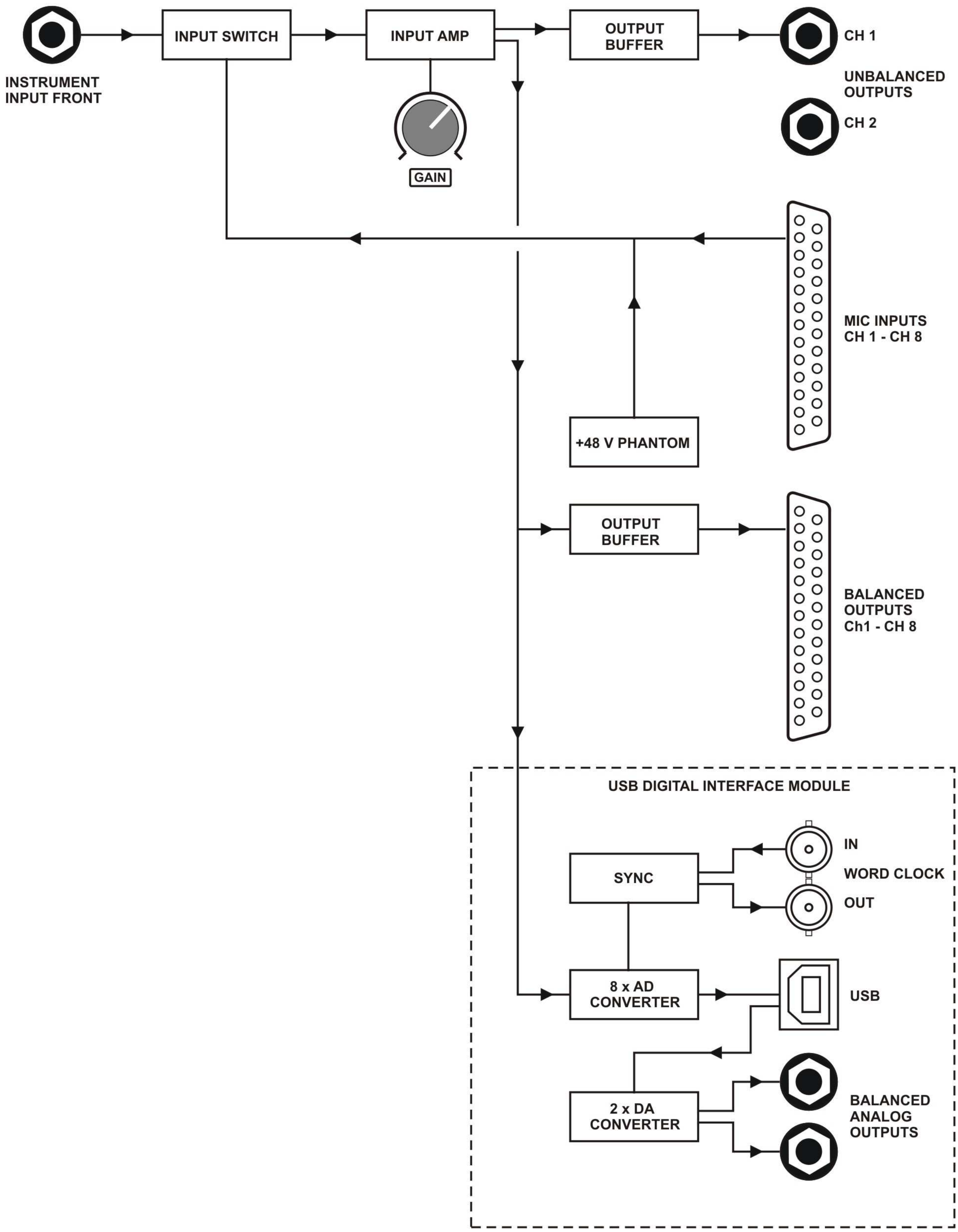
The amp has 8 separate channels with a common linear low noise Power Supply. Each channel has two inputs, a microphone input and a line input. The 8 microphone inputs are placed at the back panel and use one 25 pin D-SUB female connector. The connector follows the Tascam/Digidesign standard.

The 8 instrument/line inputs are placed at the front panel for easy access and use 1/4" standard Jack connectors

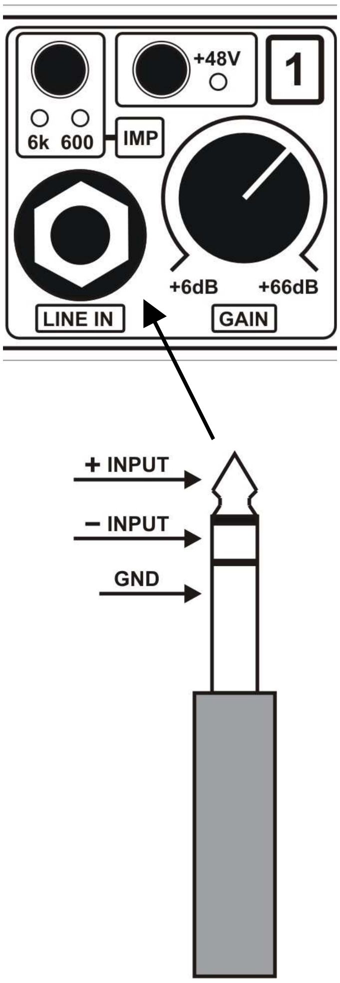
The 8 balanced line outputs are placed at the back panel and use one 25 pin D-SUB female connector, which follows the Tascam/Digidesign standard.
Additionally, channels 1 and 2 have unbalanced outputs via 1/4" standard Jack connectors.
The unbalanced and balanced outputs are always active at the same time.

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.
The mic input impedance is selected by a toggle switch at the front panel as 6 kohm or 600 ohm. The +48 Volts Phantom Power is likewise selected by a toggle switch at the front panel. LEDs indicate the selected status.

## LMA8 SIMPLIFIED SIGNAL FLOW, ONLY CHANNEL 1 SHOWN

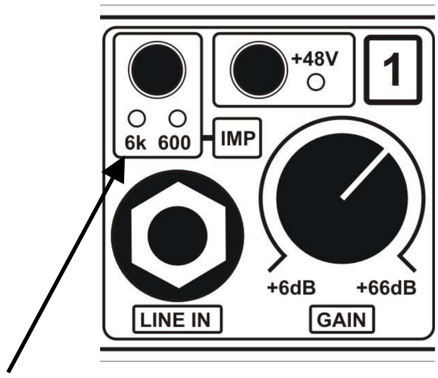


## *Operational description*




***Instrument/line input***

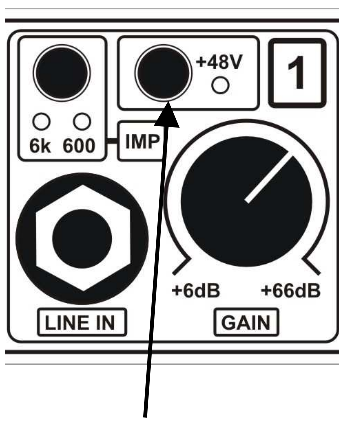
Each channel has a ¼" Jack instrument/line input on the front panel. The input is for use with instruments (like guitar, bass etc) and it can be used in a balanced as well as an unbalanced configuration.
When a Jack is plugged-in, the input circuit automatically disconnects the mic input, switches to high impedance mode and activates the line input.
Plugging in a stereo Jack enables the balanced input configuration. The tip is the +input, the ring is the -input, and the pole is the ground.
Plugging in a mono Jack enables the unbalanced input configuration. The tip is the +input and the pole is the ground.




***Mic input impedance***

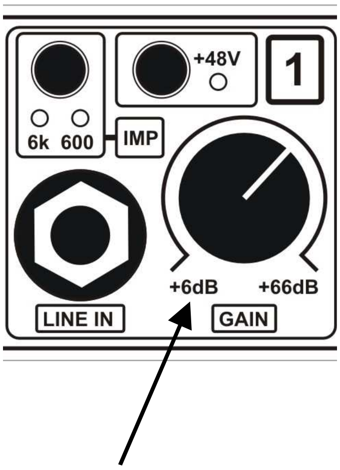
The mic input impedance is selected by pressing the toggle switch. There are two settings:
6 kohm is the preferred setting for most condenser microphones. It is the ideal setting for older high quality microphones like U47, M49 and the like. The impedance is linear up to far above the audio band, giving these vintage microphones optimum conditions for delivering their renowned detailed, open and warm sound.
600 ohms is a useful setting for dynamic microphones and non-transformer coupled ribbon microphones. The setting is also used with newer condenser microphones when a strict linear frequency response is required from the microphone, fx in acoustical measurements, industrial noise measurement etc.

*Note: Using the 600 ohm setting puts a heavier load on the microphone than does the 6 kohm setting, so a decrease of the output level from the microphone should be expected. The decrease is usually from 1 to 6 dB depending on the actual output impedance of the microphone.*




***+48V Phantom Voltage***

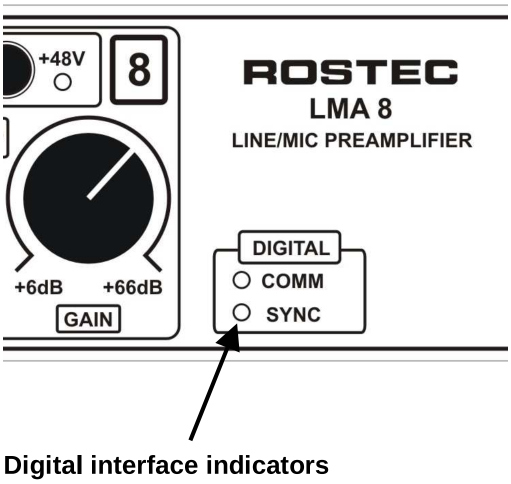
The toggle switch turns the Phantom Voltage at the mic input on and off. The LED indicates the status.




***Input gain adjust***

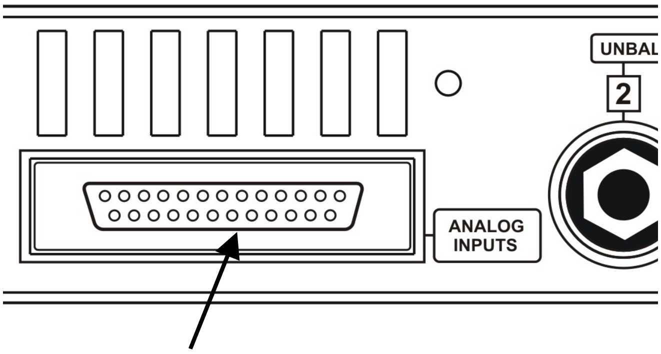
Turning the knob controls the input gain in one smooth movement. No clicks and no jumps. It works for both instrument/line input and microphone input.
The gain range is +6 to +66 dB from input to the balanced output on the D-SUB connector at the back..
The gain range is 0 to +60 dB from input to the unbalanced output on the Jack connector at the back (only channels 1 and 2).

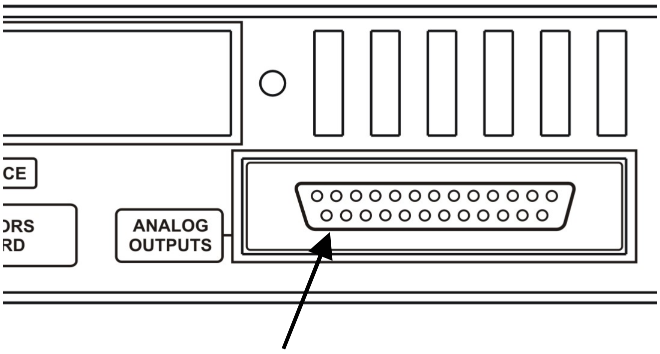
*At minimum gain setting, the Jack input and the Jack output works like a buffered pass thru, while feeding the signal to the balanced output with +6 dB gain*
*This is the traditional line driver set-up.*




**Digital interface indicators**

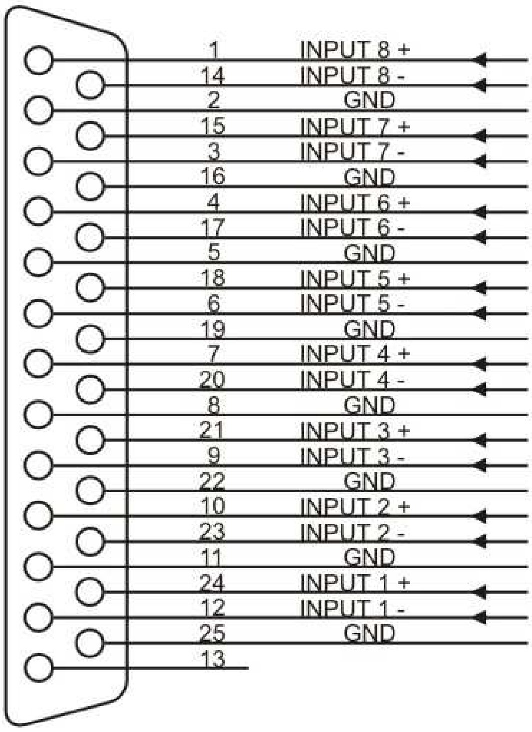
These LEDs are status indicators for the digital interface module (USB or AES). Both indicators should be steady on during normal operation.
COMM indicates a good communication link to the DAW.
SYNC indicates a solid lock to the incoming word clock.




***Analog inputs (mic inputs)***

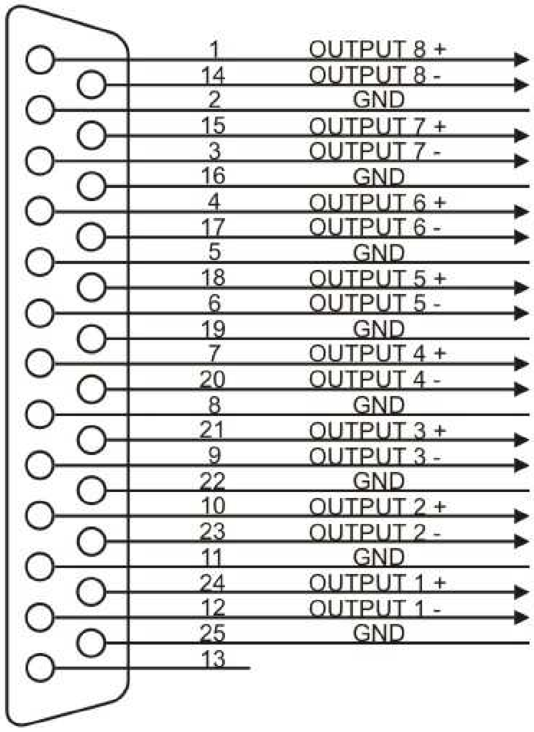
The analog input D-SUB connector is for the microphone inputs. The max input is +30dBu. The input impedance follows the setting on the front panel.
The 25 pin D-SUB connector follows the TASCAM standard as shown below.
Observe that channel 8 is at the beginning of the pin numbers and channel 1 is at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

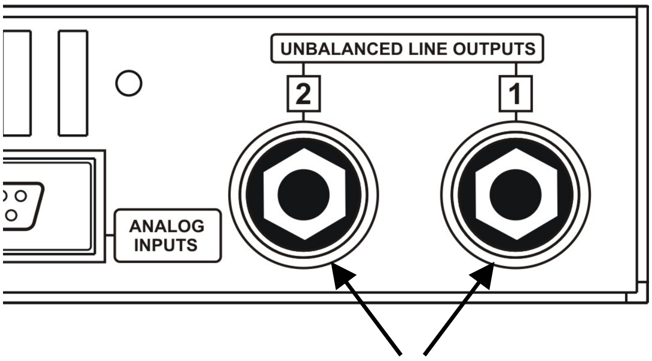



***Analog outputs (line outputs)***

The analog output D-SUB connector is for the balanced line outputs. The output buffers are transformerless balanced floating configuration. For unbalanced operation, the negative output must be shorted to ground Max output is +30 dBu in balanced mode.  Max output is +24 dBu in unbalanced mode.

Like the input connector, the 25 pin D-SUB connector follows the TASCAM standard

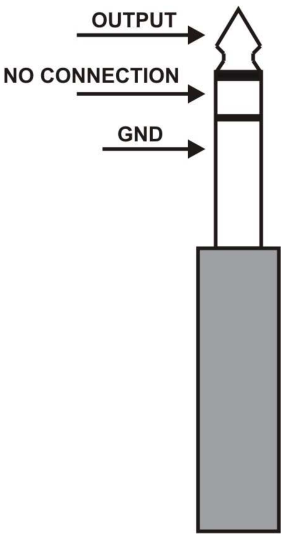


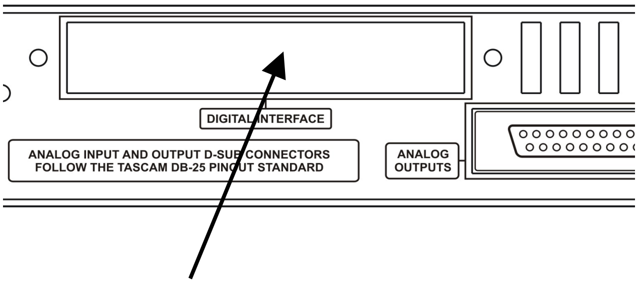



***Unbalanced outputs (line outputs)***

The two ¼" Jack connectors are for the unbalanced line outputs. These outputs are intended for use with instrument amplifiers (guitar, bass etc)
The max output level is +24 dBu.

The connectors are meant for standard mono ¼" Jack plugs. Plugging-in a stereo Jack will leave the ring floating, using a balanced cable as an unbalanced cable.



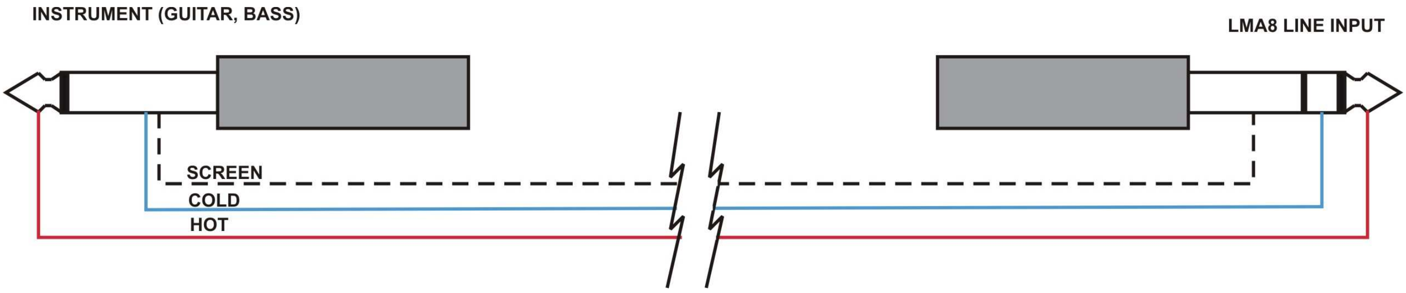



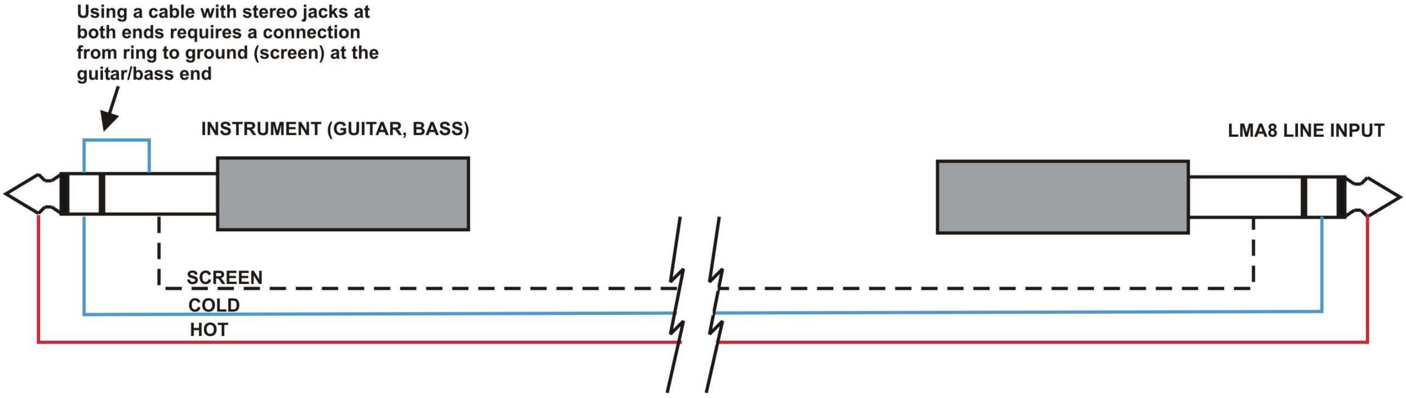

***Digital interface slot***

This is the slot for the optional digital interface module, USB or AES. The interface module is installed by opening op the box and fastening the module with screws and connecting it with ribbon cables.

The procedure and the technical description of the USB and AES modules can be found in separate documents.

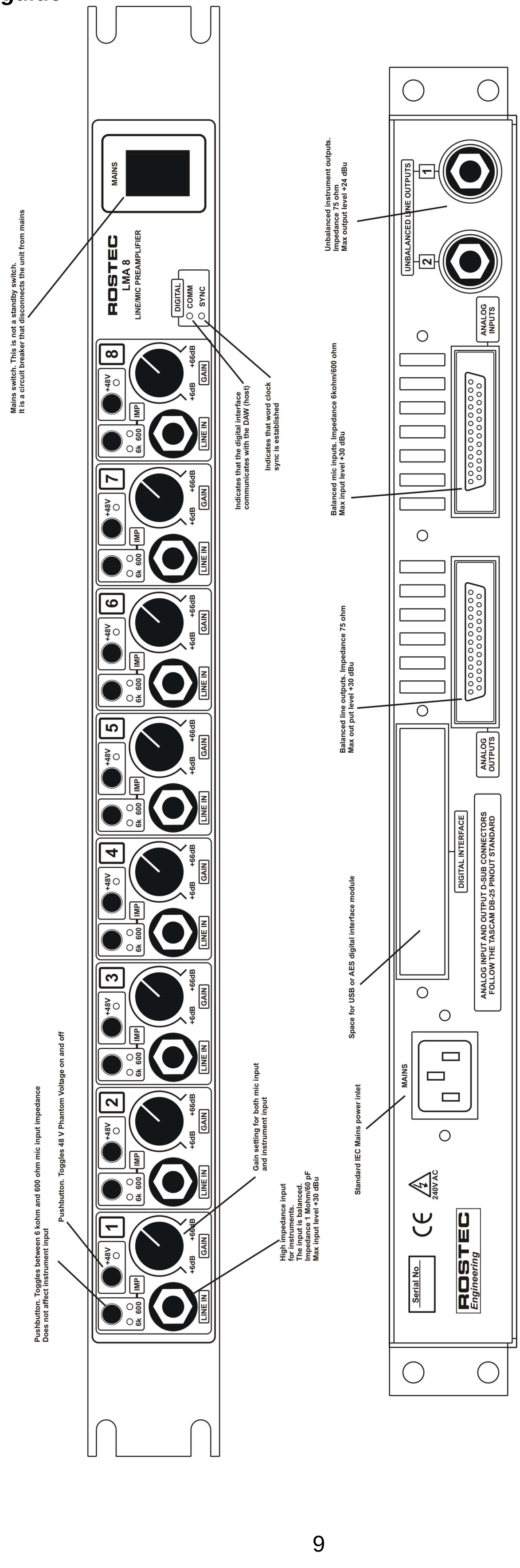
***Remote ground connection***




The balanced instrument/line input of the LMA8 opens op 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 end, 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 from the cable is affected.

***Front and back panel quick guide***



***Mechanical and electrical specifications***

Dimensions: Width 19 inch, height 1U (44 mm), depth 225 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

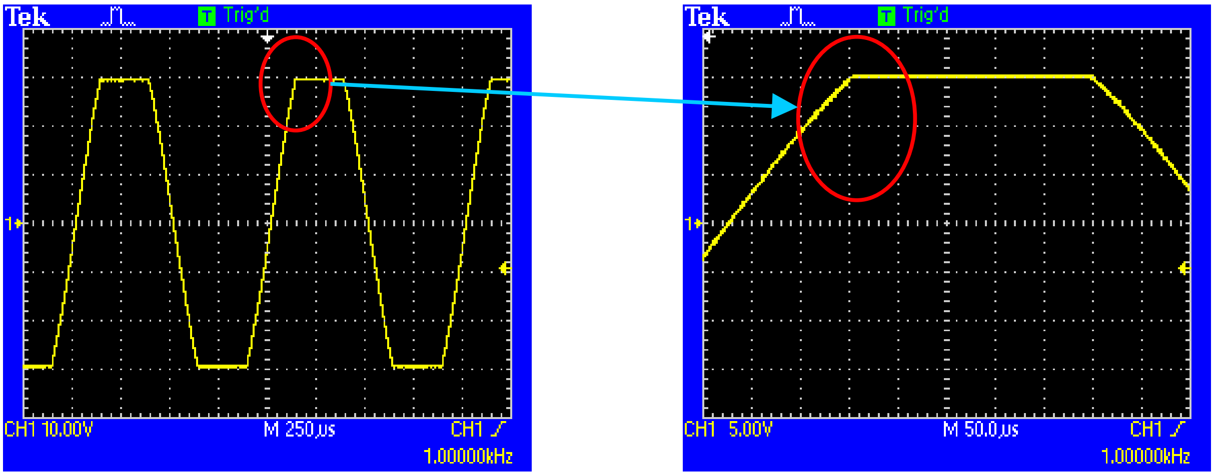
Mic input impedance: 6 kohm or 600 ohm
Line input impedance: 1 Mohm/60 pF
Max input level balanced, mic or line: +30 dBu
Max input level unbalanced, line: +24 dBu
Max output level, balanced output: +30 dBu
Max output level, unbalanced output +24 dBu
Gain, input to balanced output: +6 dB to +66 dB
Gain, input to ch1/ch2 unbalanced output: 0 dB to +60 dB
Input noise: -134 dBu (A weighted, 22 Hz - 22 kHz)
Input noise: -131 dBu unweighted
Frequency response: 10 Hz - 100 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, 10 dB gain

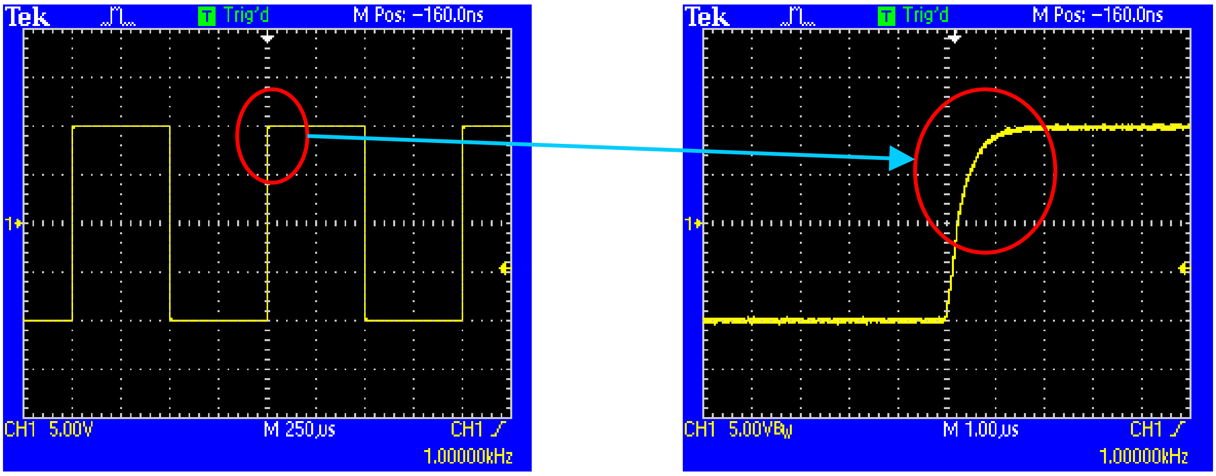
# Technical section

### *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 6 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. 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 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***

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

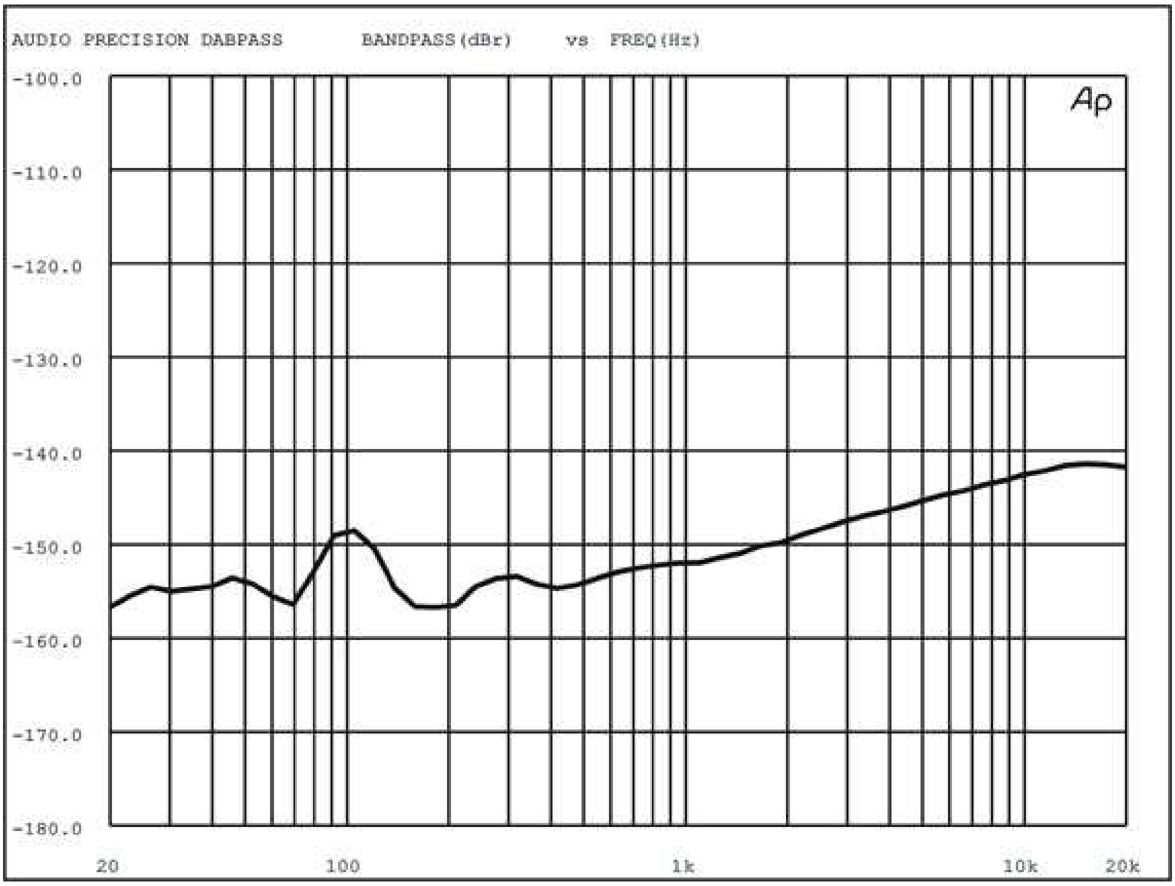
*But not to worry; LMA8 has a mechanism in place to prevent such unwanted HF feedback.*

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) | 210 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 |

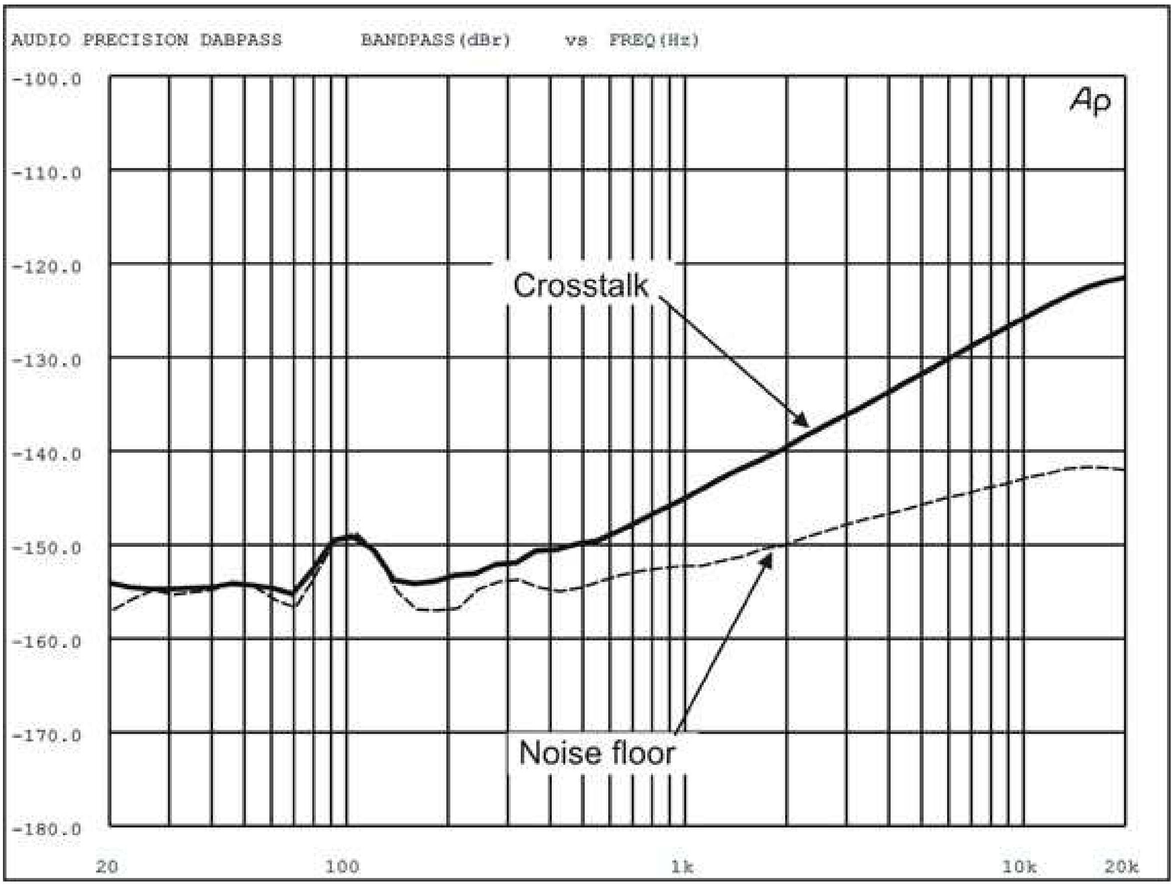
The change of the frequency response is not audible and it does not affect internal loop gain, noise figures or distortion figures.

***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 (+66 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 -148 dBu
The rms sum of the noise from 22 - 22 kHz is -134 dBu A weighted, or -131 dBu unweighted.




**Crosstalk**

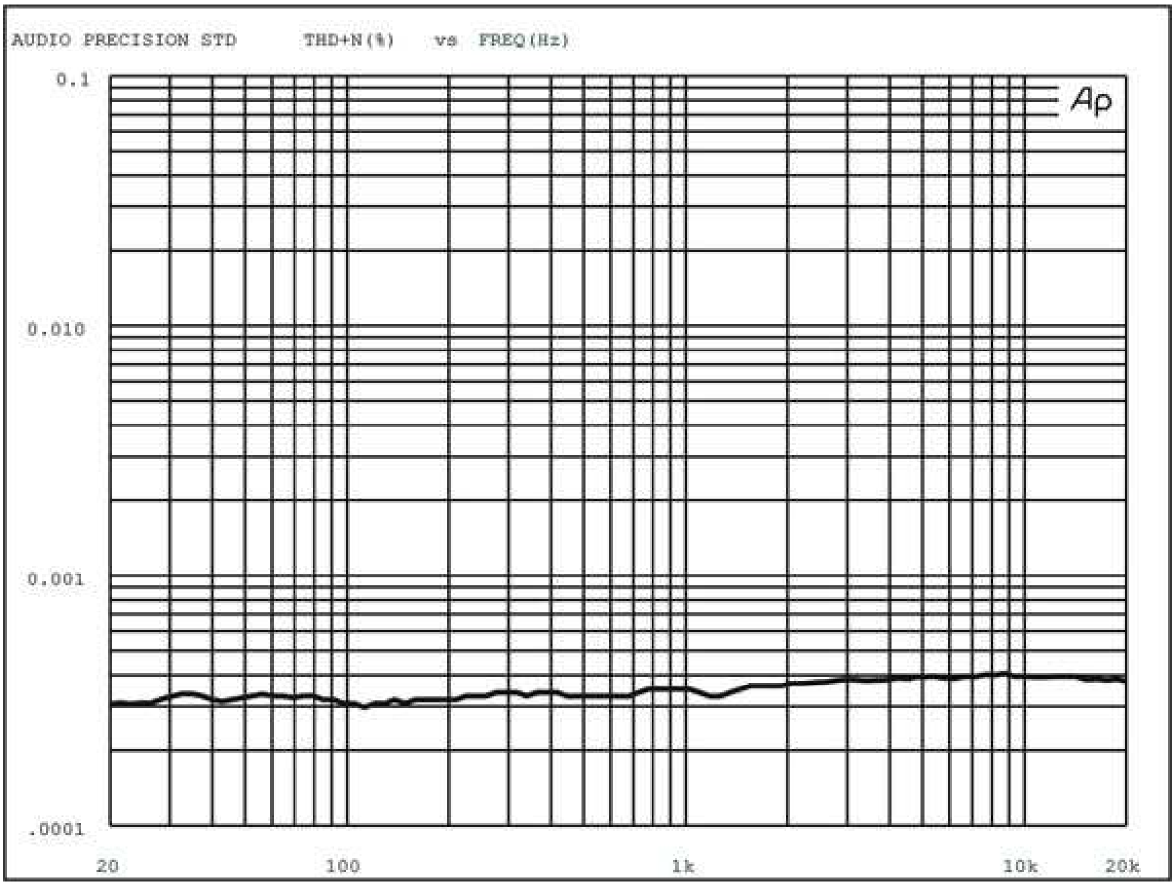
The graph shows crosstalk between adjacent channels. A +20 dBu signal was sent to channel 1, gain setting +6 dB. The output level of channel 1 was +26 dBu.
The input of channel 2 was terminated by 150 ohms, and the signal level was measured at the output of channel 2.

The graph shows virtually no crosstalk below 100 Hz, which indicates that there is no ground current transfer.
The crosstalk increases at higher frequencies, but is kept below -120 dBu. This indicates a small capacitive signal transfer internally between components.

*Channel 1 is blasting at +20/+26 dBu and channel 2 sees only -120 dBu at the input. This clearly demonstrates the benefits of the balanced architecture.*

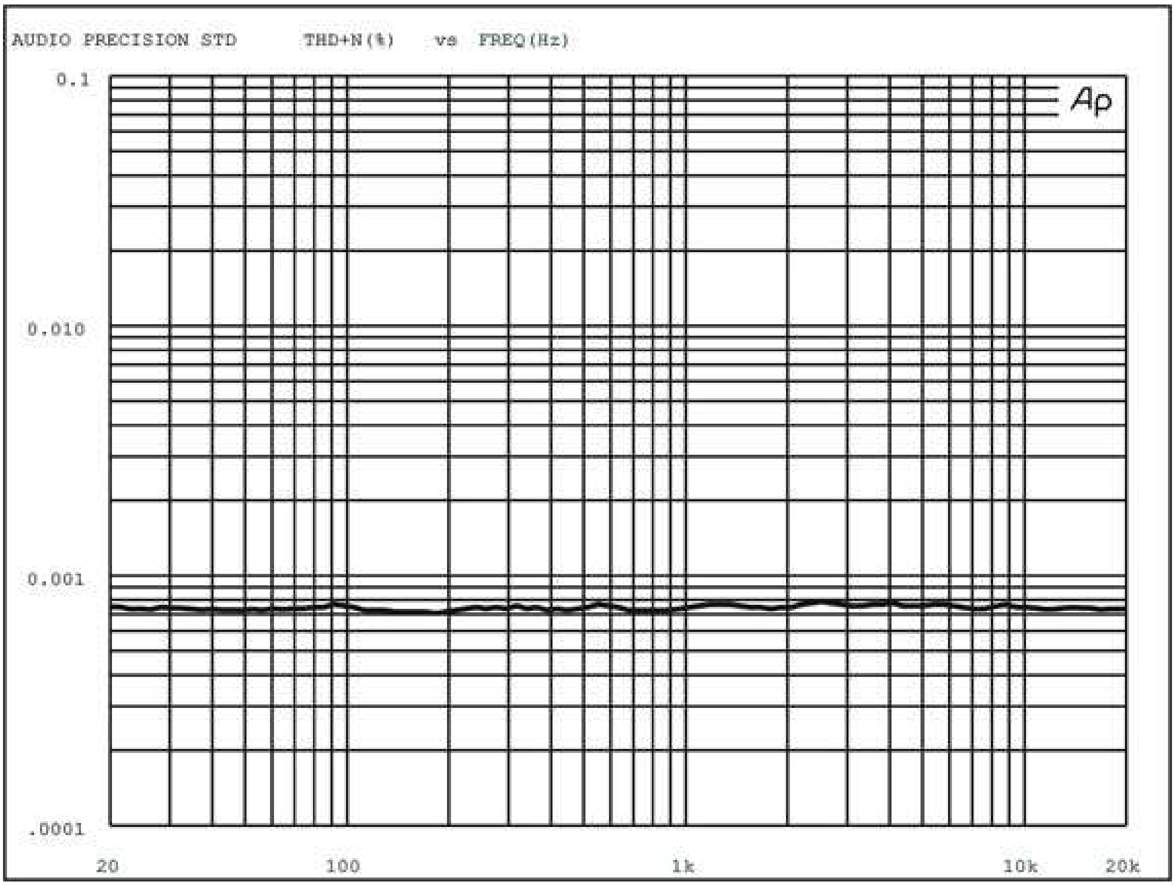
***Classical Distortion Analysis***

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 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.



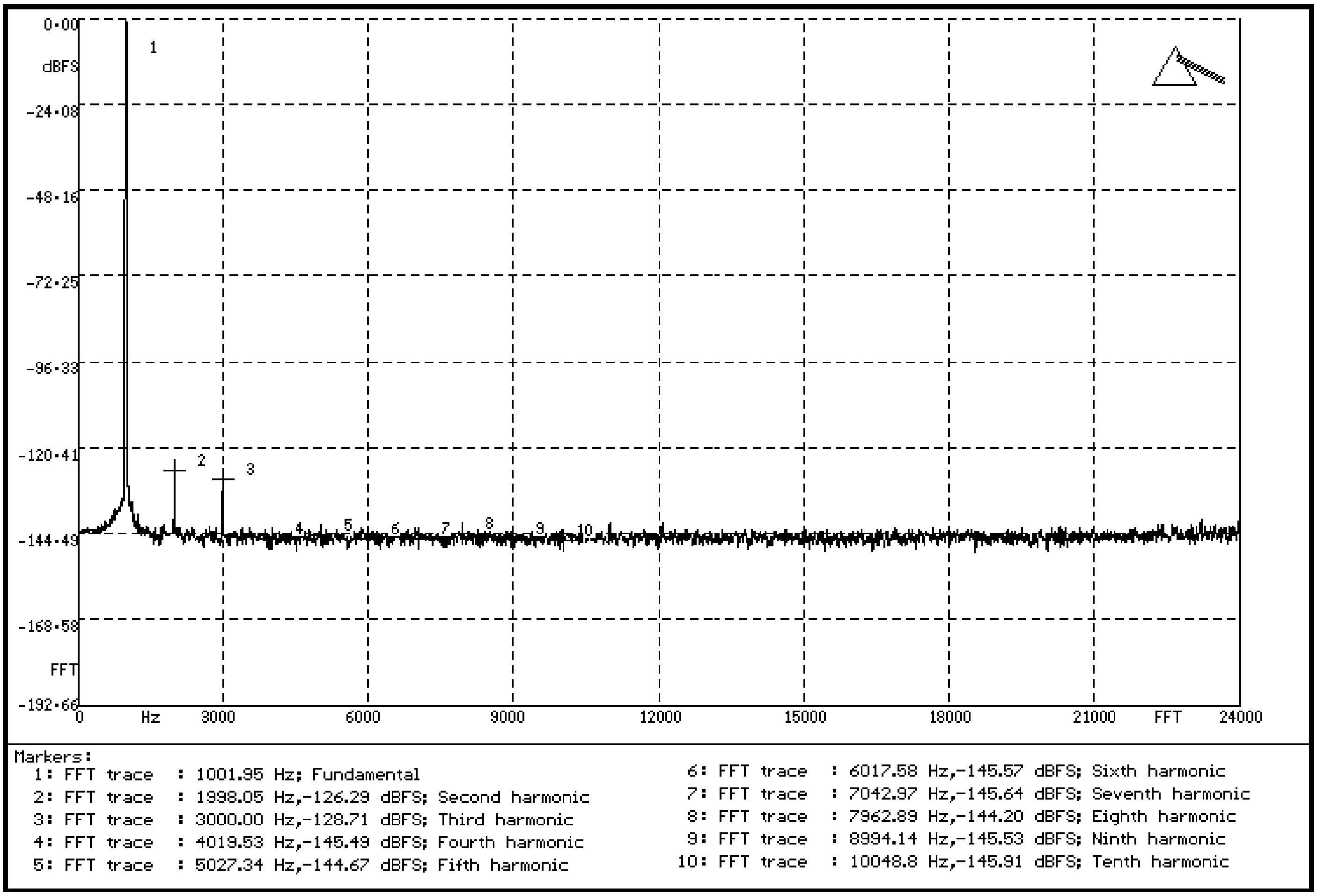

**Distortion + noise at +30 dB gain**

The THD + N is below 0.0008 % in the whole audio range. The measured result is 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.

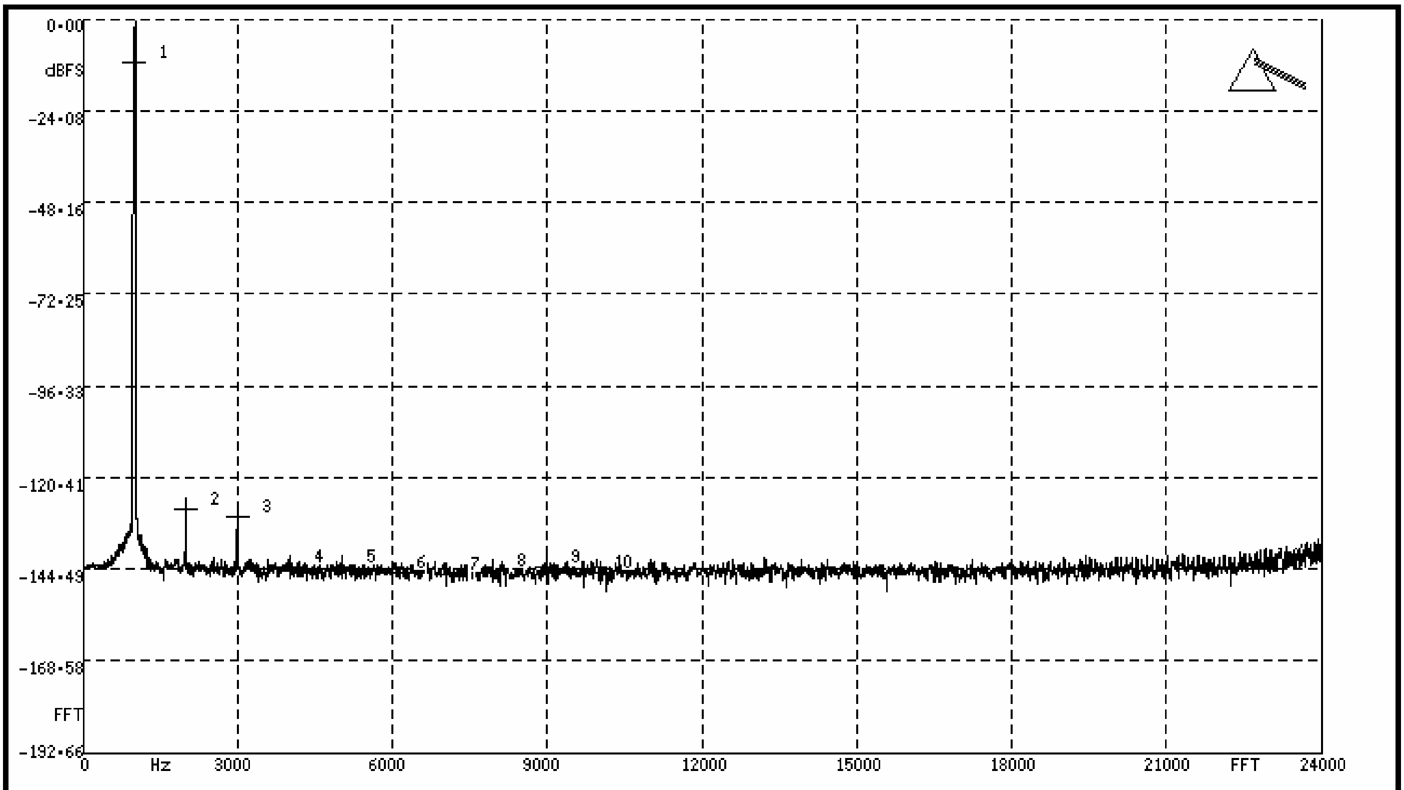
***FFT distortion analysis***

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

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 %



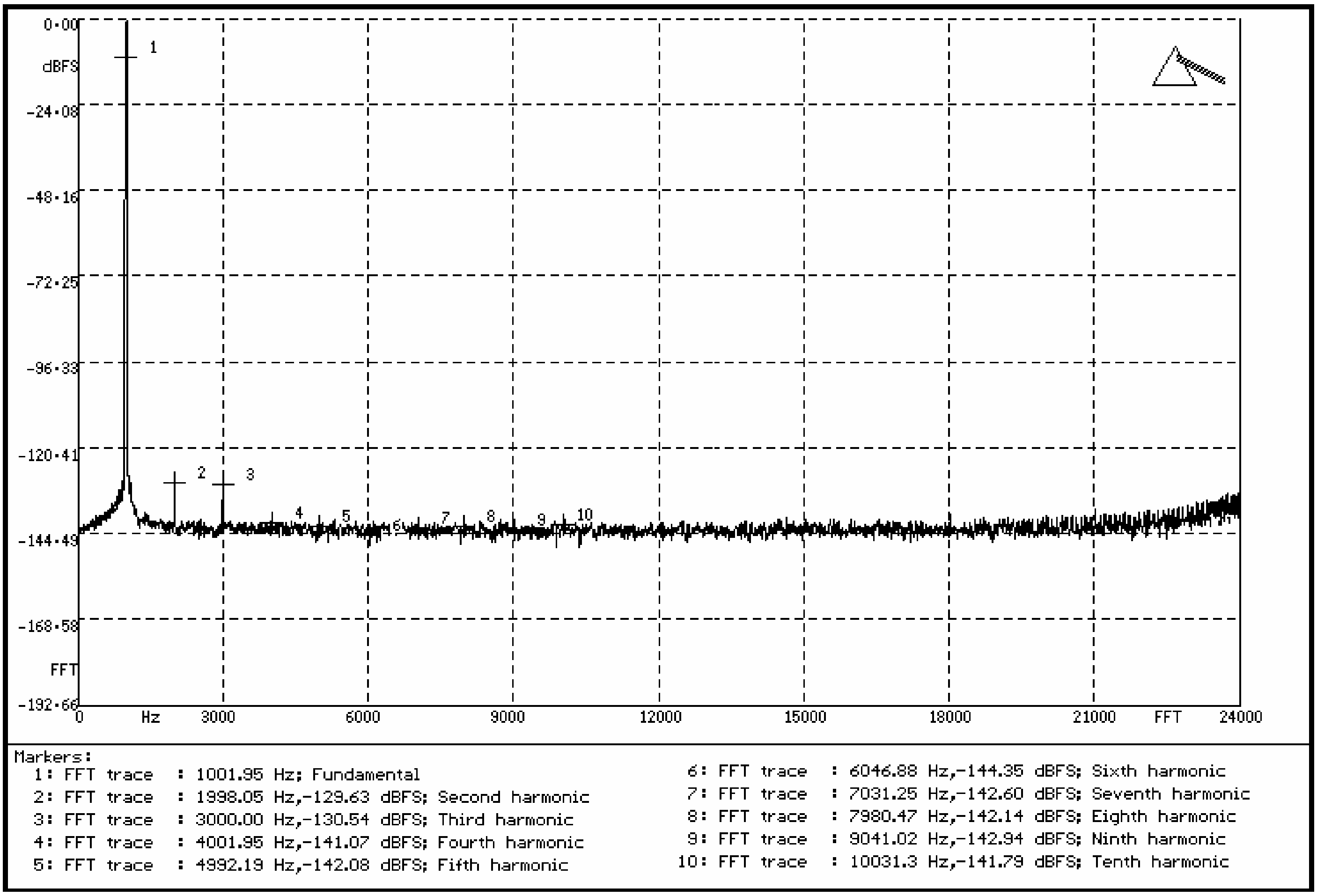

**Gain +10 dB, frequency 1 kHz. The rms sum of all distortion components (THD) is 0.00013 %**



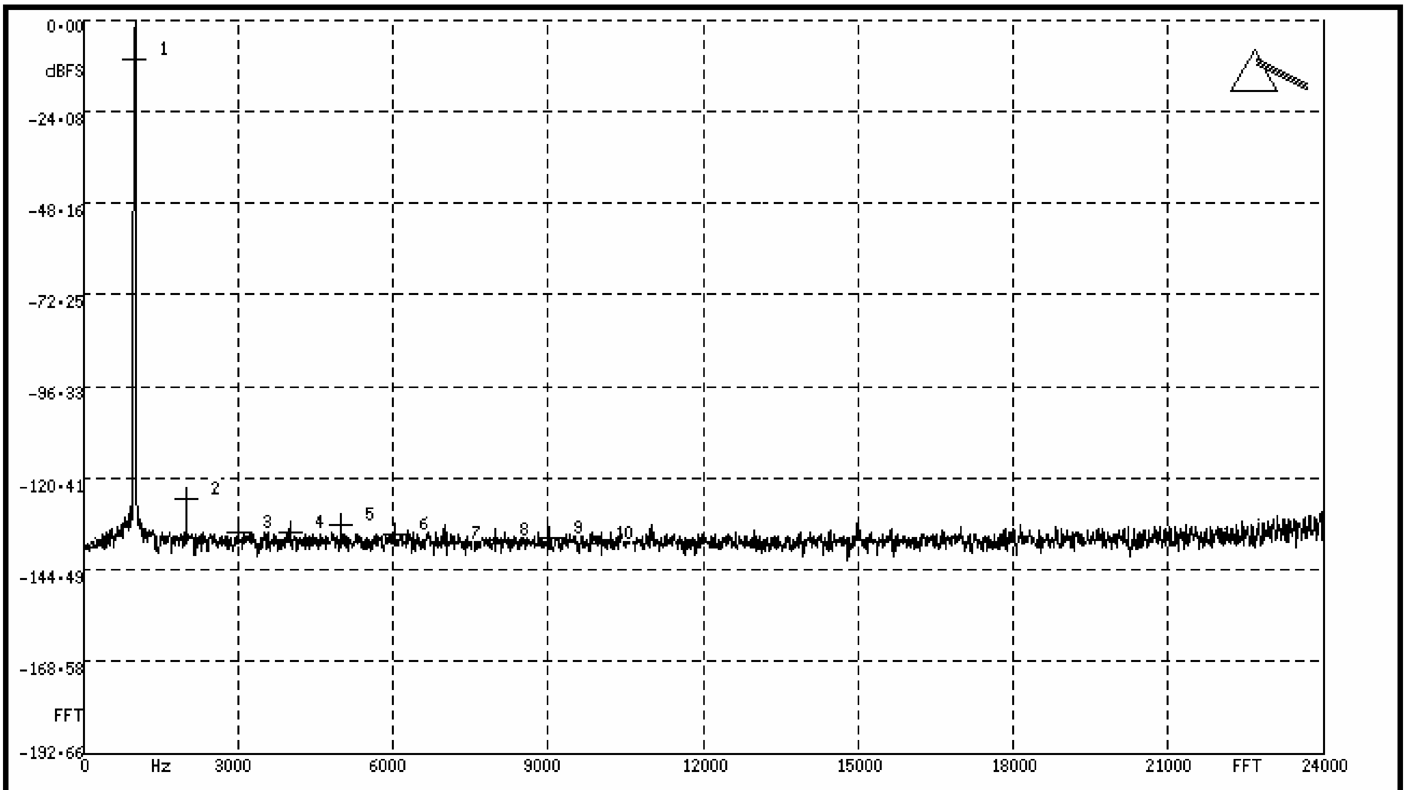

Markers:

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 %**



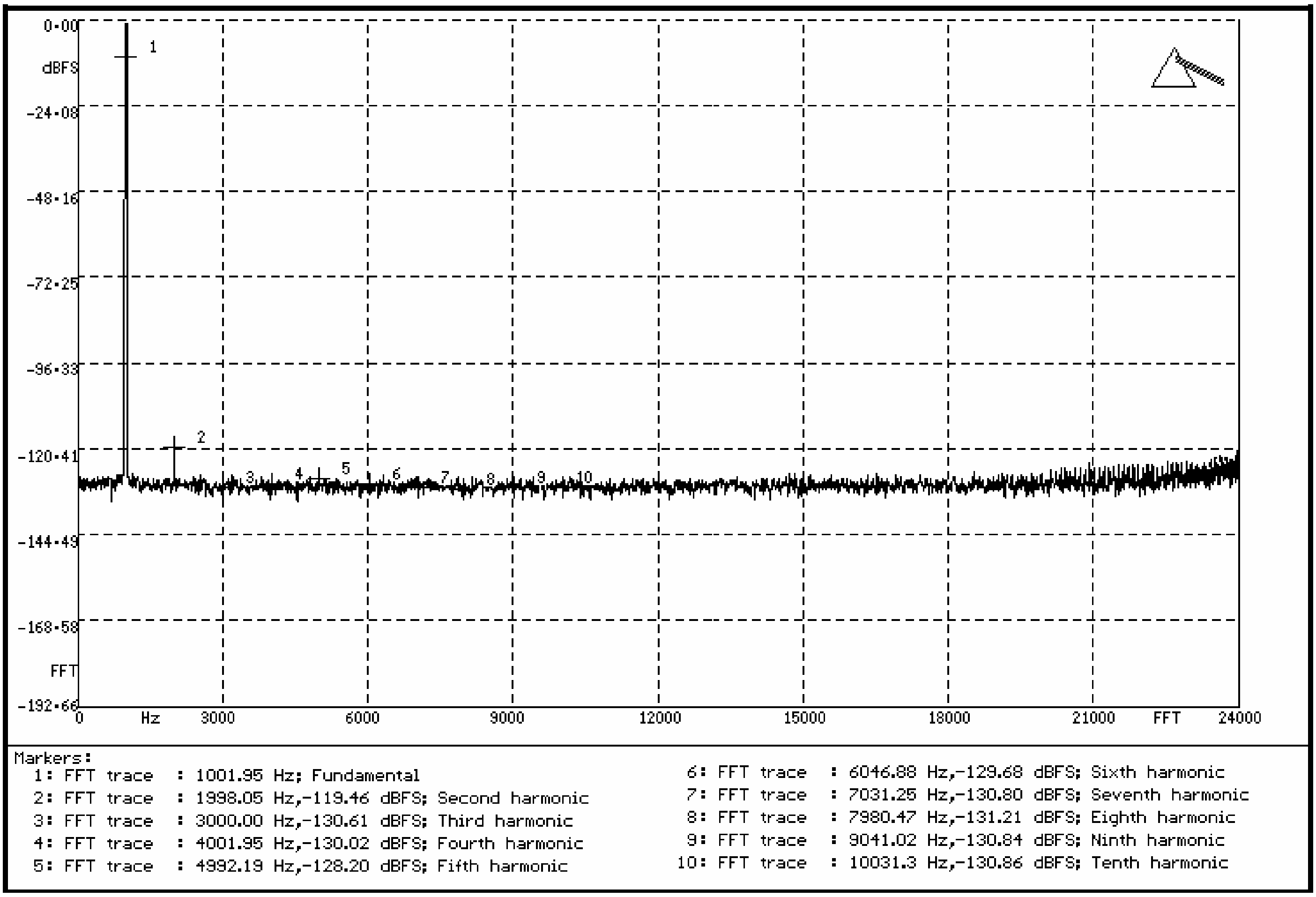

**Gain +30 dB, frequency 1 kHz. The rms sum of all distortion components (THD) is 0.00015 %**



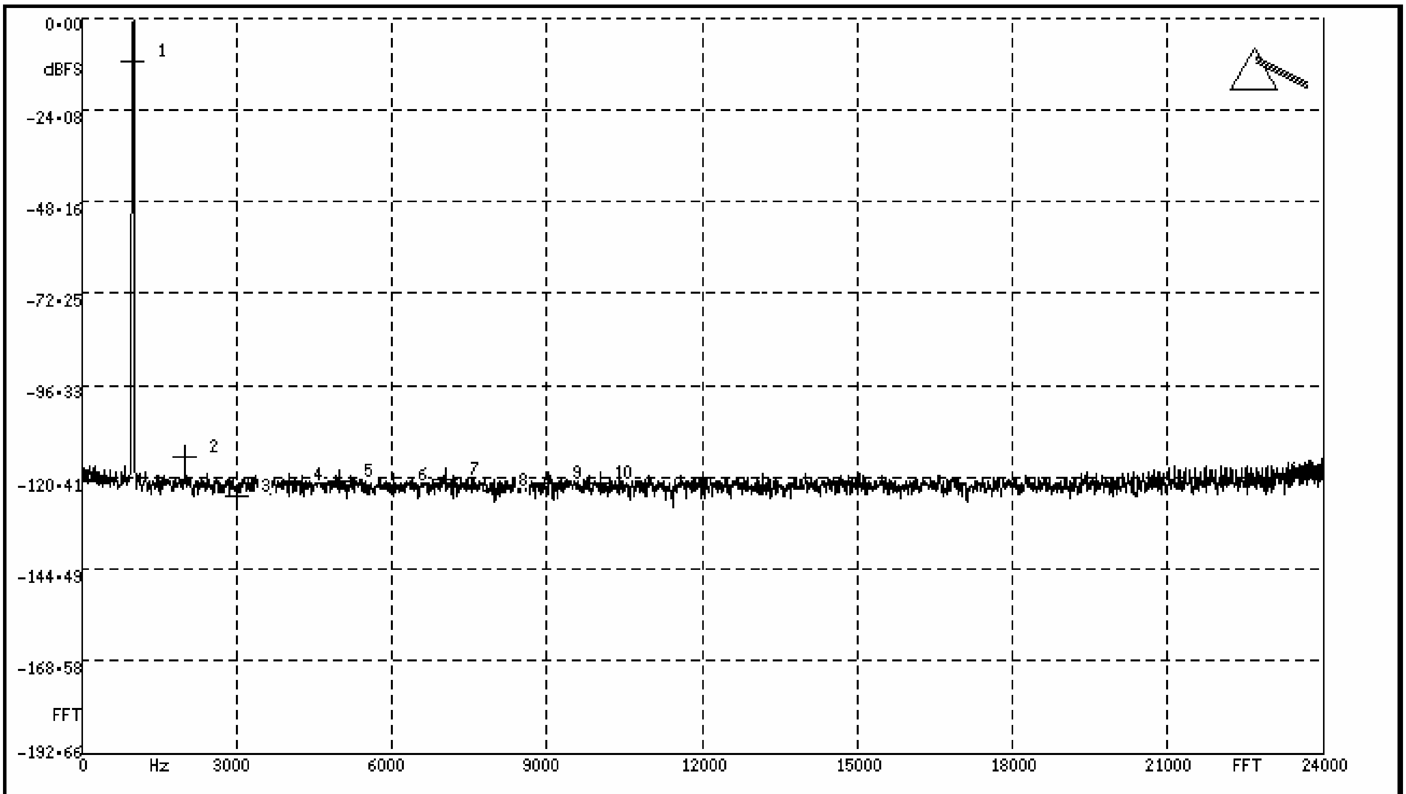

Markers:

1: FFT trace : 1001.95 Hz; Fundamental
2: FFT trace : 1998.05 Hz,-125.28 dBFS; Second harmonic
3: FFT trace : 3000.00 Hz,-134.44 dBFS; Third harmonic
4: FFT trace : 4001.95 Hz,-134.01 dBFS; Fourth harmonic
5: FFT trace : 4992.19 Hz,-132.34 dBFS; Fifth harmonic
6: FFT trace : 6046.88 Hz,-134.62 dBFS; Sixth harmonic
7: FFT trace : 7031.25 Hz,-136.99 dBFS; Seventh harmonic
8: FFT trace : 7980.47 Hz,-136.37 dBFS; Eighth harmonic
9: FFT trace : 9041.02 Hz,-135.51 dBFS; Ninth harmonic
10: FFT trace : 10031.3 Hz,-137.17 dBFS; Tenth harmonic

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




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




Markers:

1: FFT trace : 1001.95 Hz; Fundamental
2: FFT trace : 1998.05 Hz,-114.57 dBFS; Second harmonic
3: FFT trace : 3000.00 Hz,-124.89 dBFS; Third harmonic
4: FFT trace : 4001.95 Hz,-121.76 dBFS; Fourth harmonic
5: FFT trace : 4992.19 Hz,-120.92 dBFS; Fifth harmonic
6: FFT trace : 6046.88 Hz,-121.95 dBFS; Sixth harmonic
7: FFT trace : 7031.25 Hz,-120.42 dBFS; Seventh harmonic
8: FFT trace : 7980.47 Hz,-123.32 dBFS; Eighth harmonic
9: FFT trace : 9041.02 Hz,-121.52 dBFS; Ninth harmonic
10: FFT trace : 10031.3 Hz,-121.34 dBFS; Tenth harmonic

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