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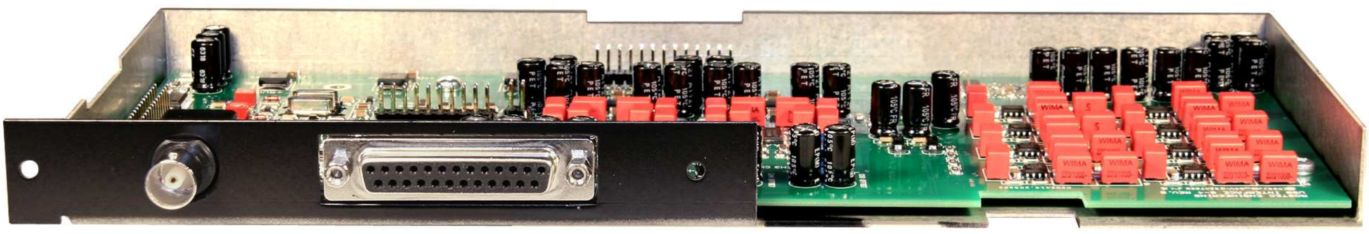
AES Module for LMA8 Mic/Line Preamplifier



AES Module, AM88

8 channel analog in, 8 channel digital out

For ROSTEC LMA8 preamplifier

Revision 5, August 08-2016

General description

Technical section

AES Module for LMA8 Mic/Line Preamplifier

Features

- Fully compliant transformer balanced AES3, IEC-60958 transmitter Interface
- Bit-perfect AES data transfer
- 8 channels of high-end 192 kHz, 24-bit, 118 dB analog to digital converters
- Combination of JFET and BIPOLAR technology for exquisite sonic performance
- Impressive low distortion specifications
- Fully balanced analog architecture
- Taps the audio signal directly at the preamp circuit for shortest possible signal path.
- Exceptional low jitter on-board clock generator
- Word sync input has a "sweet spot" detector with signal clean-up and clock regeneration
- Works as a slave to any AES compliant system, auto detecting the sampling frequency via word clock
- Inputs and outputs are ESD protected to 23 kV, IEC 61000-4-2 and 15 A surges, IEC 61000-4-5.

General description

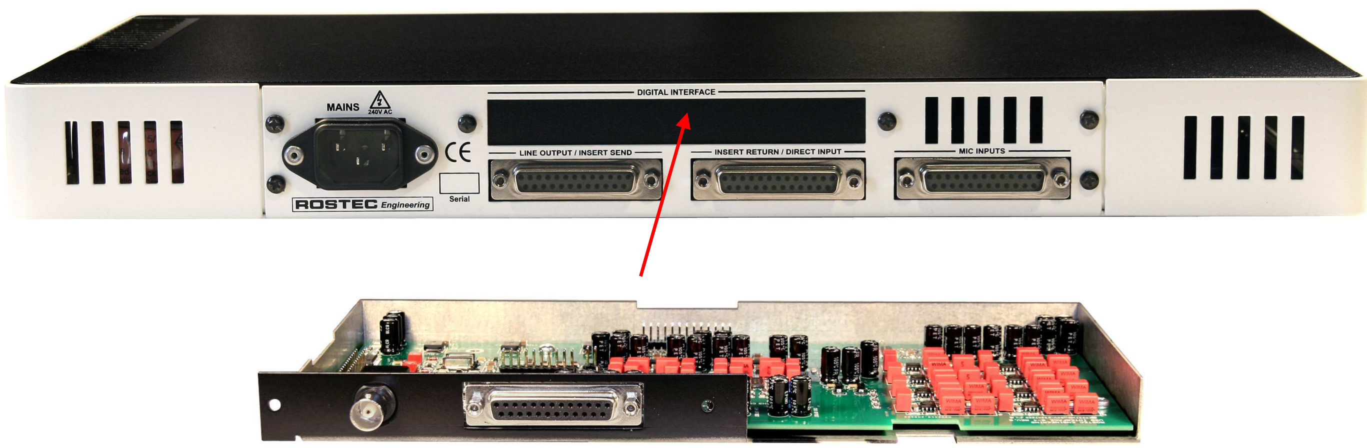
The AES Module is a fully self-contained 8 channel AD converter interface for the LMA8 analog preamplifier.
It features 8 channels of high-end 192 kHz 24 bit AD converters, and 4 fully transformer balanced AES stereo compatible transmitter interfaces. (8 channels)
The analog circuits are state of the art using high quality top range components, and all analog filters are of minimum phase design with strictly controlled impulse response
The module works as a slave interface, detecting the sampling frequency automatically via the incoming word clock. The actual sample rate for the module follows the frequency of the incoming clock.

When no word clock is available, the module shuts down and enters a low power mode. When a word clock is fed to the module, the internal clock generator immediately locks to the incoming clock and wakes up the module.
The clock generator is able to synchronize to all the standard word clock frequencies (44.1, 48, 88.2, 96, 176.4 and 192 kHz).
The generator performs an efficient signal clean-up of the incoming clock, providing a perfect system clock for the AD converters and the AES encoders (see technical section).

Installation

The module is normally installed into the LMA8 chassis at the factory. Installation of the module in the field is not recommended, but is possible for a skilled technician. Contact ROSTEC for details.

When properly installed, the mechanical chassis parts create a closed metal cage, creating an effective screen that physically isolates all digital signals from the line/microphone circuits of the LMA8.



Analog inputs

During normal operation, the analog inputs of the module are tapped directly at the outputs of the preamp circuits of the LMA8 as balanced signals.
When the insert-point switch of any channel at the front panel of the LMA8 is pressed, the corresponding channel input of the module becomes accessible from the direct input/insert return D-SUB connector at the back panel of the LMA8.

Digital outputs (send to DAW)

The 8 digital audio signals from the AD converters are converted to 4 serially encoded AES audio streams, and sent to a DAW or any AES compliant system via the standard 25-pin female D-SUB connector at the back panel of the module.
(rem: 1 AES output contains 2 audio channels)

The output signals are transformer balanced and 110 ohm. The rise and fall times are extremely fast, providing a very clear and well defined signal for the receiving end. The pin-out follows the YAMAHA digital standard

About overload, clipping, headroom etc

The AES module works in perfect harmony with the LMA8 analog preamp when it comes to headroom and clipping.

At Digital Full Scale (dBFS), the analog level of the outputs at the back panel of the LMA8 is +20 dBu.
As the output buffers have a max output level of +30 dBu, there is always extra headroom even at digital full scale. Thus output buffer clipping of the LMA8 is totally irrelevant.

Input clipping of the LMA8 mic/line inputs is equally irrelevant, because the headroom of the input circuit is even higher than the output buffer headroom.
When the signal level from the LMA8 mic preamp is high enough to bring about digital clipping, the mic preamp still has 16 dB of extra headroom!

Analog clipping is simply not an issue in a LMA8/AES module configuration.

*The benefits of this “**comfy zone**” philosophy are evident. The LMA8, with the AES module installed, has an exceptional airy, open and clear sound that surprises even the most seasoned engineers and producers.*

Further, to protect the AD converters against overload when digital clipping occurs, a hard limit protection circuit is in place. The protection circuit is active just above the digital clipping point, and does NOT interfere with the audio signal in any way.

"Actually, the clipping sounds kind of great on bass and drums, provided that your recording software doesn't puke when it happens"

(See technical section for clipping characteristics)

Word Clock Generator

The internal word clock generator controls all the digital clocks used by the AES module.
The actual sampling frequency is set by the generator, based on the frequency of the incoming word clock at the BNC connector at the back panel of the module
The clock generator is able to synchronize to all the standard word clock frequencies (44.1, 48, 88.2, 96, 176.4 and 192 kHz).

When no word clock is available, the module is kept in a low power mode.
When an incoming word clock is detected, the internal crystal reference oscillator is enabled, and the internal clocks are fed to the AD converters and AES encoders, after which normal operation commences.
The generator 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, keeping the circuit momentarily running at the last received clock frequency by using an internal fly-wheel.
If no clock is received before fly-wheel time-out, the generator shuts down the module.
If the clock returns before time-out, the generator softly glides from the internal reference to the incoming clock, and resumes normal operation without any gaps or interruption of the digital audio signal.

Shorter gaps in the incoming sync are efficiently absorbed that way. The soft gliding back and forth is sufficiently well damped in order for the AES module to absorb this without any degradation of sound.

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 output clock.

An input clock with high level of jitter gets the treatment too. The 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 as the reference point. The result is a word clock with typically less than 0.1 nsec RMS jitter, and system clocks (master clock, serial clock, data clock etc.) with almost immeasurable jitter in the range of a few picoseconds.

The solid jitter performance and high stability of the clock generator provide an ideal environment for the AD converters and AES encoders.
This is one of the basic reasons for the exceptional sonic quality of the AES module.

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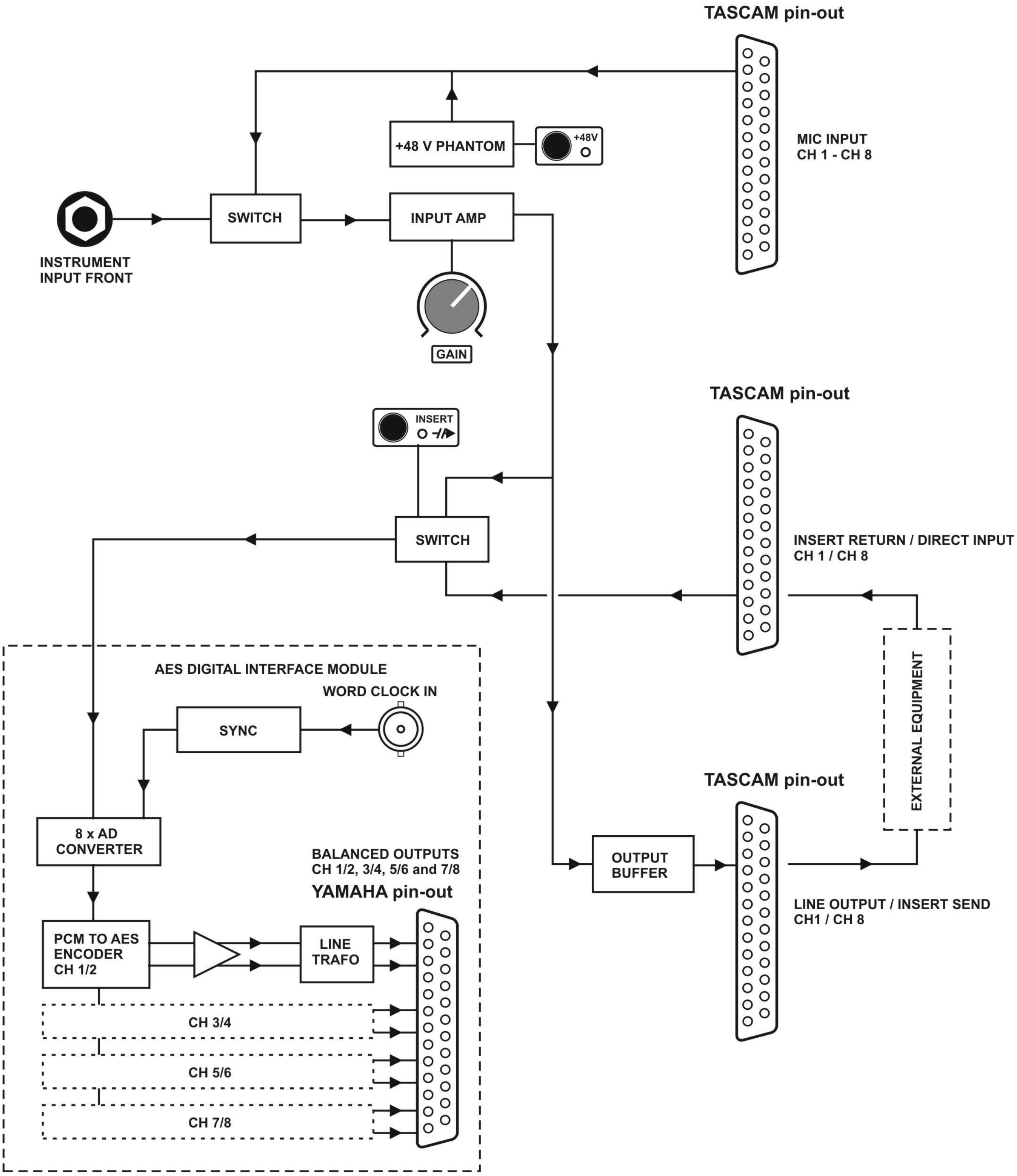
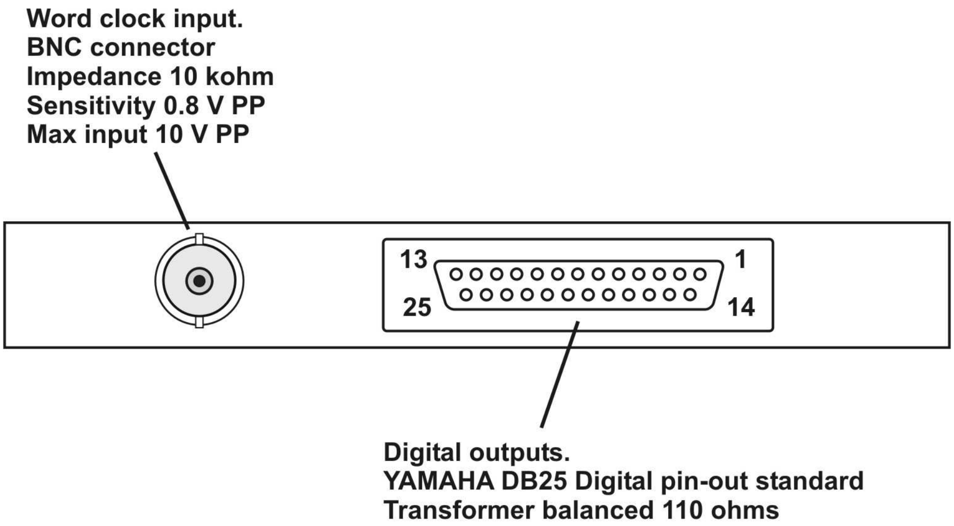
LMA8 WITH AES MODULE

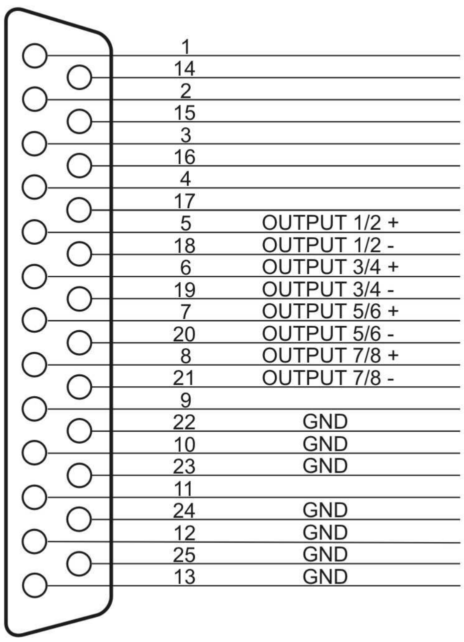
SIMPLIFIED SIGNAL FLOW, ONLY ONE CHANNEL SHOWN

Pin number connections are just illustrative



Connector panel quick guide





Mechanical and electrical specifications

Dimensions: Width 267 mm, height 20 mm, depth 112 mm
Weight: 1.0 kg
Power requirements: +8-10 V 330 mA, +15 V 110 mA, -15 V 110 mA
ESD: Protected to 23 kV, IEC 61000-4-2 and 15 A surge, IEC 61000-4-5

Word Clock generator:
Accuracy, internal crystal: 1 ppm at 25 degC, factory adjusted
Temperature stability: +/-2 ppm from 0 degC to +70 degC
Ageing 2 ppm pr. year
Word clock frequencies: 44.1 k, 48 k, 88.2 k, 98 k, 176.4 k, 192 k
Internal crystal oscillator jitter: typical 2 ps RMS
PLL capture range: +/-50 ppm
Lock time approx: 0.3 sec

Word Clock input impedance:	10 kohm
Word Clock input level:	Min 0.8 V PP, Max 10 V PP

Digital output:
AES3 according to IEC-60958
Output impedance 110 ohm, transformer balanced
4 stereo channels
Output 25 pin D-SUB, Yamaha Standard

AD converter:
8 channels

Resolution:	Max 192 kHz 24 bit
Dynamic range:	118 dB, limited to 116 dB by op-amp buffer noise
THD+N:	0.0004 % @ 1kHz (see THD+N measurements)
THD:	0.00015 % @1 kHz (see FFT analysis)

Analog:

Analog input level:	+14 dBu internally, +20 dBu externally from LMA8 back plane
Crosstalk adjacent channel:	-126 dB at 1 kHz

Technical section

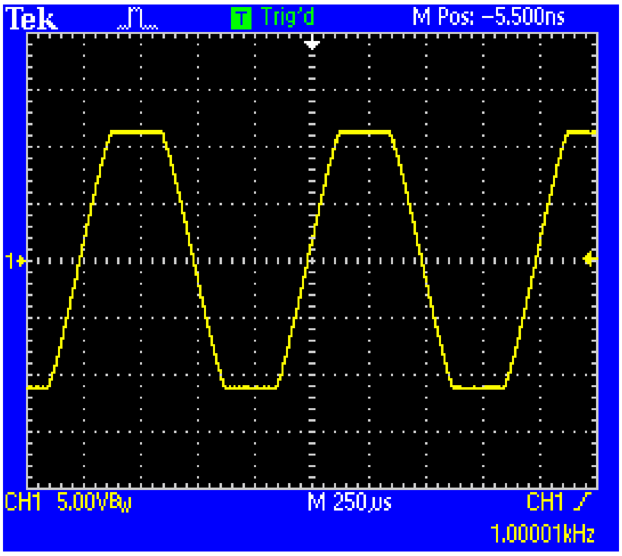
Clipping characteristics

Digital clipping has always had a bad name, and with good reason too. Many AD converters sound like a total system crash when they are brought into clipping. Various manufacturers have employed elaborate solutions to compensate for this, like input compressors and "soft limiters" etc. In reality, this is just swapping one bad thing for another.
Rostec uses a different approach. The AES module is designed to accept input overload as a normal condition, and as a result the clipping characteristics is just like any other high quality analog circuit.

To illustrate this, a sine tone was sent to the AD converter, sent on to the DAW via AES (plug-in card), and sent back from the DAW via AES to a reference DA. The signal was measured at the output of the DA with an oscilloscope. Ergo, it is an analog in/analog out measurement - a "real world" measurement. Below are some scope snapshots of the clipping characteristics.

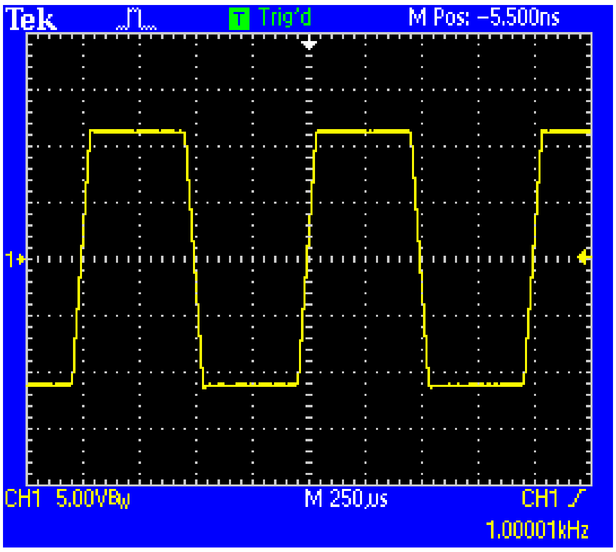
+3 dB overload.
Sampling frequency: **192 kHz.**
Analog frequency: 1 kHz.
Measurement bandwidth: 200 MHz.

The output shows moderate clipping. Notice how regular the sine is cut. No overshoot and no recovery delay. This kind of clipping is barely audible. It just adds a little spice to the sound.



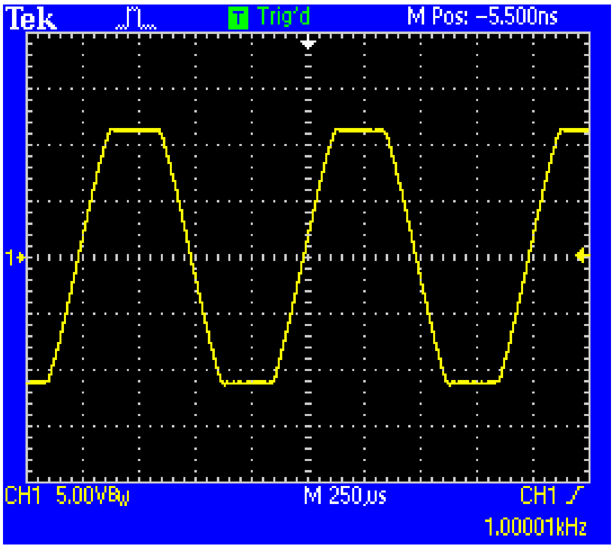
+12 dB overload.
Sampling frequency: **192 kHz.**
Analog frequency: 1 kHz.
Measurement bandwidth: 200 MHz.

The output shows severe clipping. The input level is far above the output capability of a standard professional console. Again, no overshoot and no recovery delay. And more important, no phase reversal or chopping-up of the signal!
This kind of clipping is clearly audible, but it actually sounds great on drums and bass.



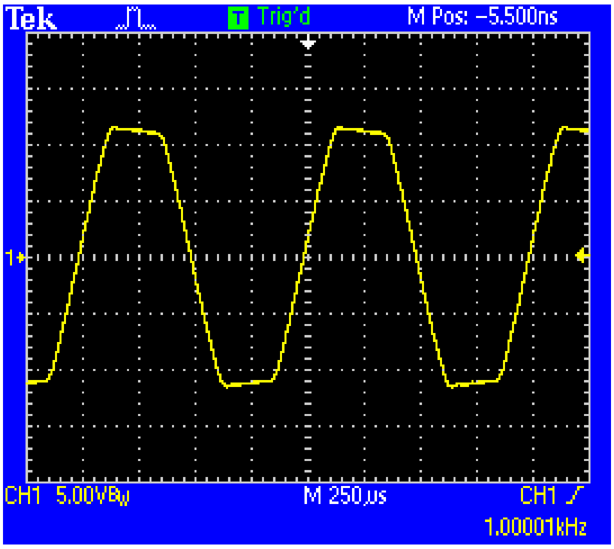
+3 dB overload.
Sampling frequency**: 96 kHz.**
Analog frequency: 1 kHz.
Measurement bandwidth: 200 MHz.

The output shows clipping that is very close to the clipping characteristics at 192 kHz sampling frequency. Very regular cutting, no overshoot and no recovery delay. The clipping is barely audible



+3 dB overload.
Sampling frequency**: 48 kHz.**
Analog frequency: 1 kHz.
Measurement bandwidth: 200 MHz.

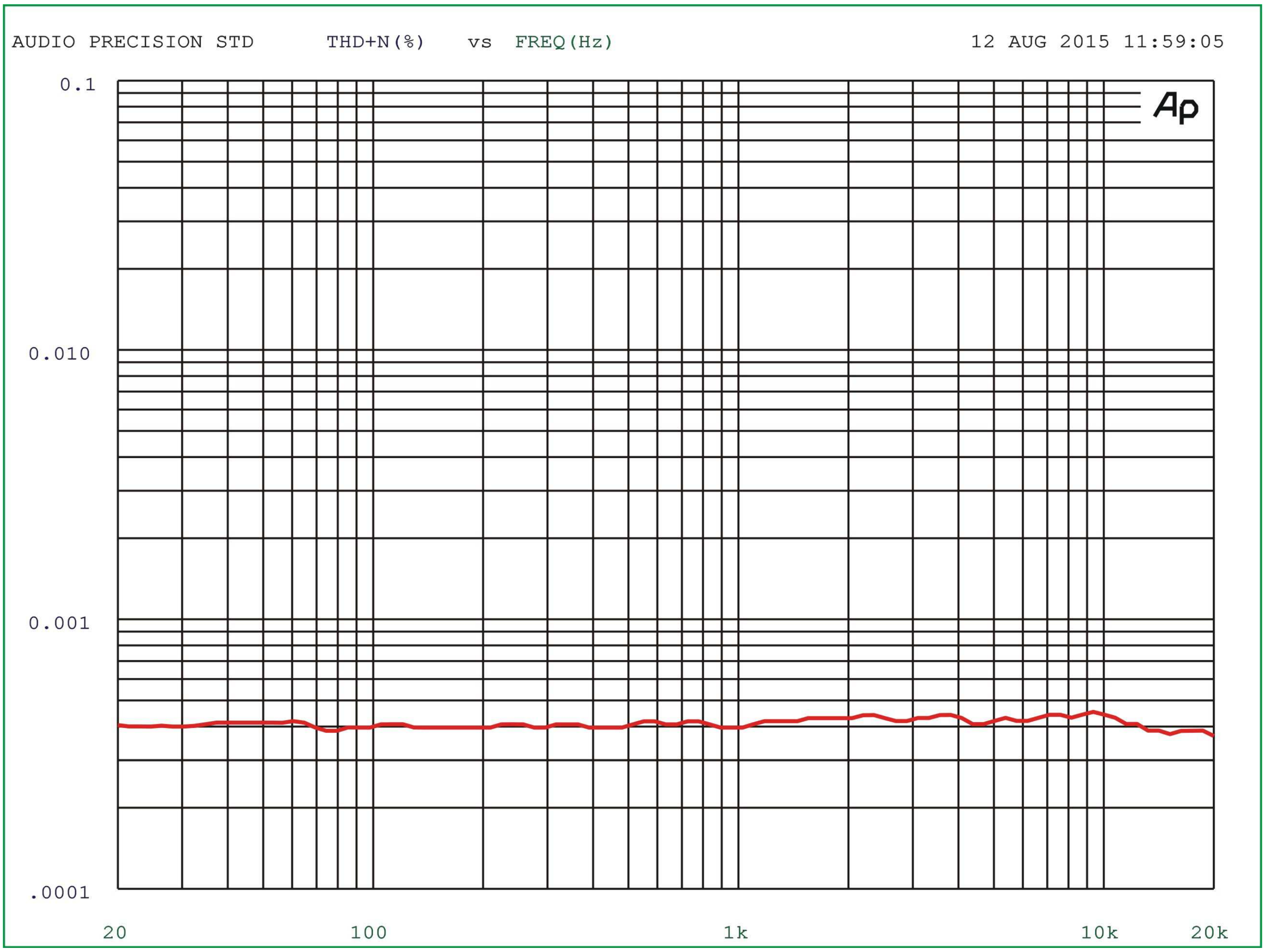
At 48 kHz sampling frequency, the clipping shows a less regular cut. At first glance it looks like lo-cut filtering, but it is actually due to a small ringing from the 100 dB digital anti-aliasing filter in the converter chip. There is absolutely nothing unusual in this. This is how digital filters work.
The clipping is still just barely audible.



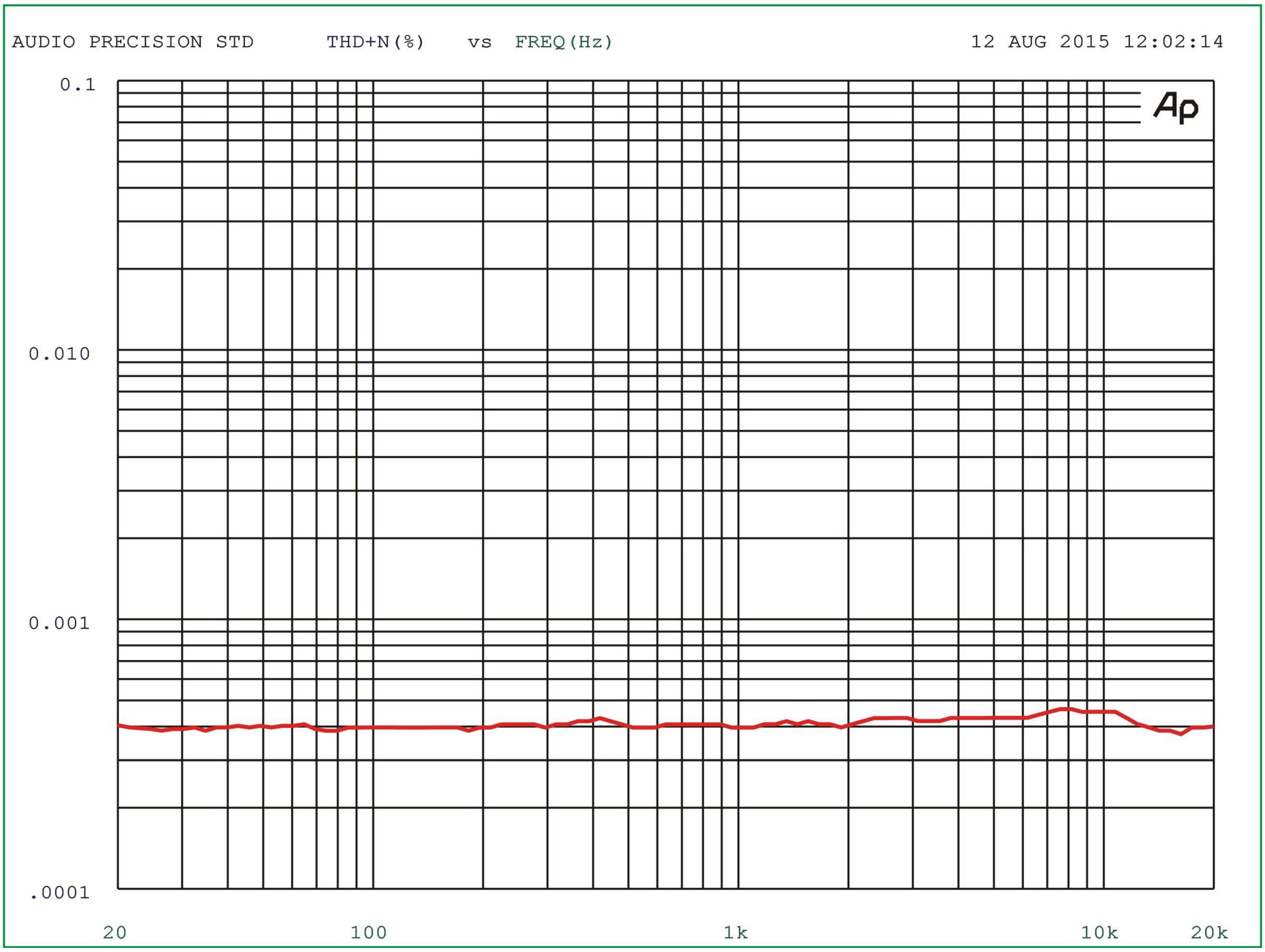
Distortion characteristics

The distortion test was performed as a traditional THD+N test. An analog signal at 1 dB below Digital Full Scale, sweeping from 20 Hz to 20 kHz, was sent to the module; and the resulting output was measured with an Audio Precision analyzer as the RMS sum of all distortion signals PLUS all noise contributions within the full audio spectrum.

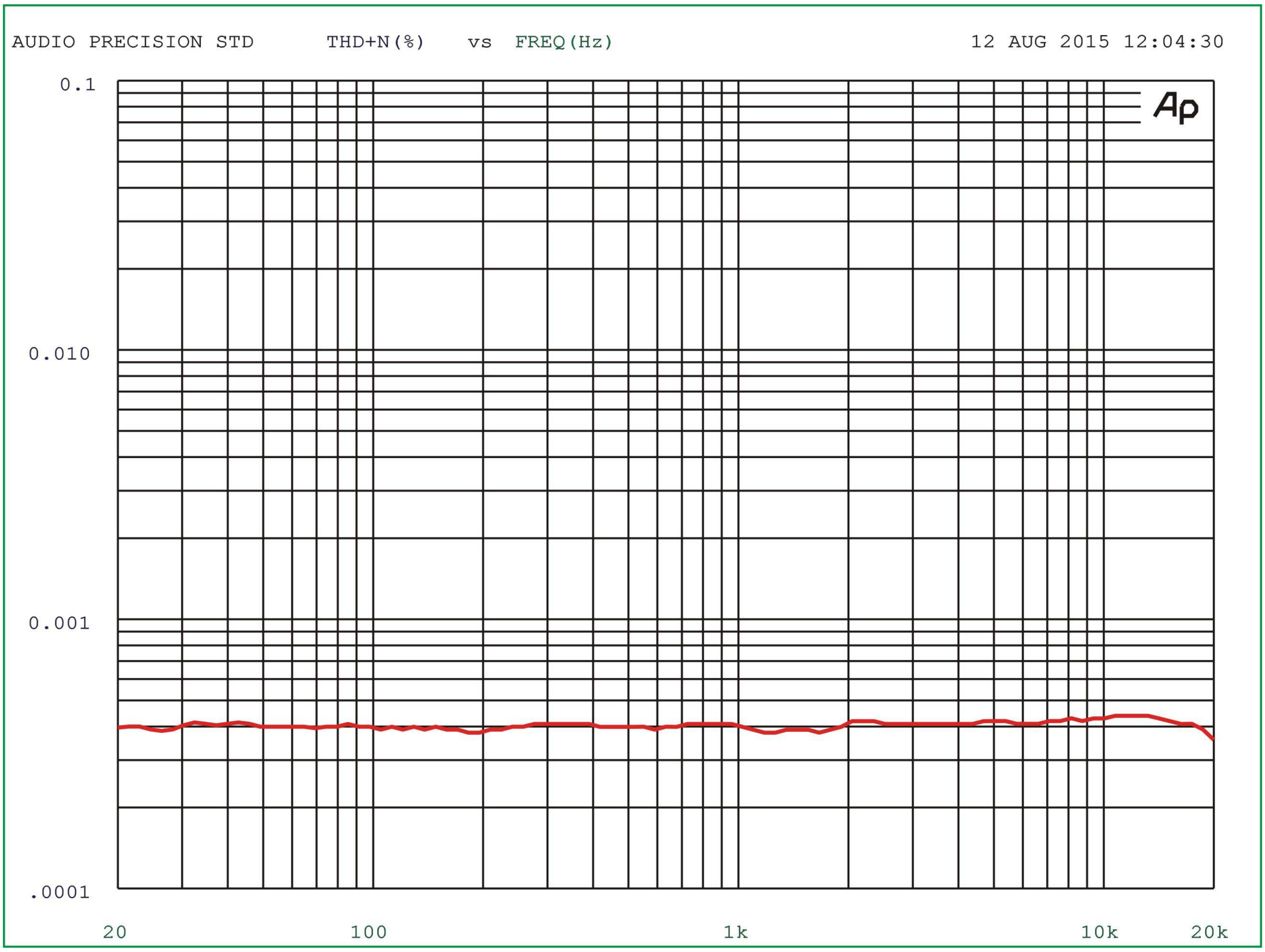
Below are the results of distortion + noise plotted versus frequency. Individual performance plots are shown for the standard sampling frequencies 44.1, 48, 88.2, 96, 176.4 and 192 kHz.



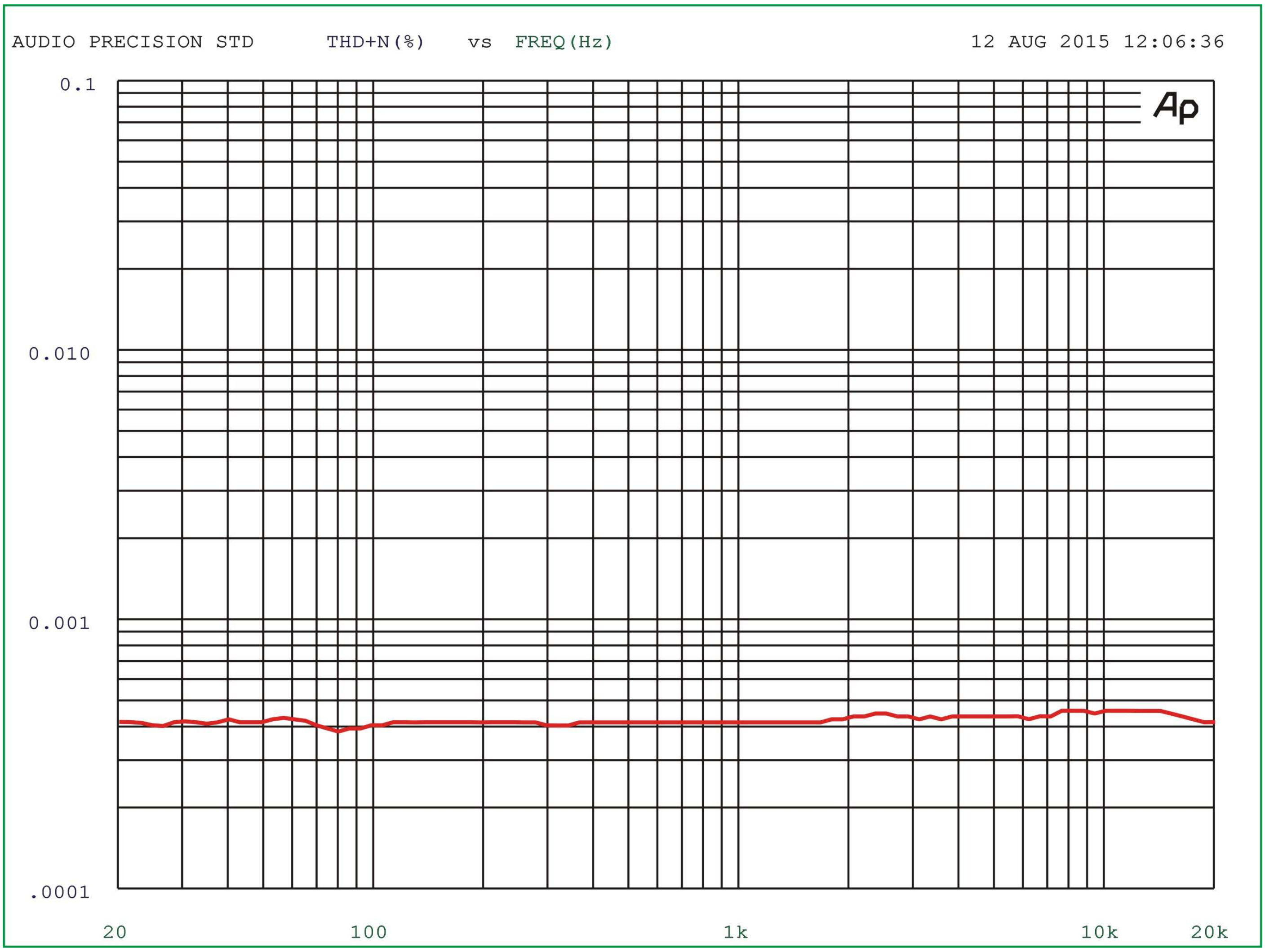
Distortion (THD+N) 20 Hz - 20 kHz @ -1.0 dBFS, sampling frequency 44.1 kHz



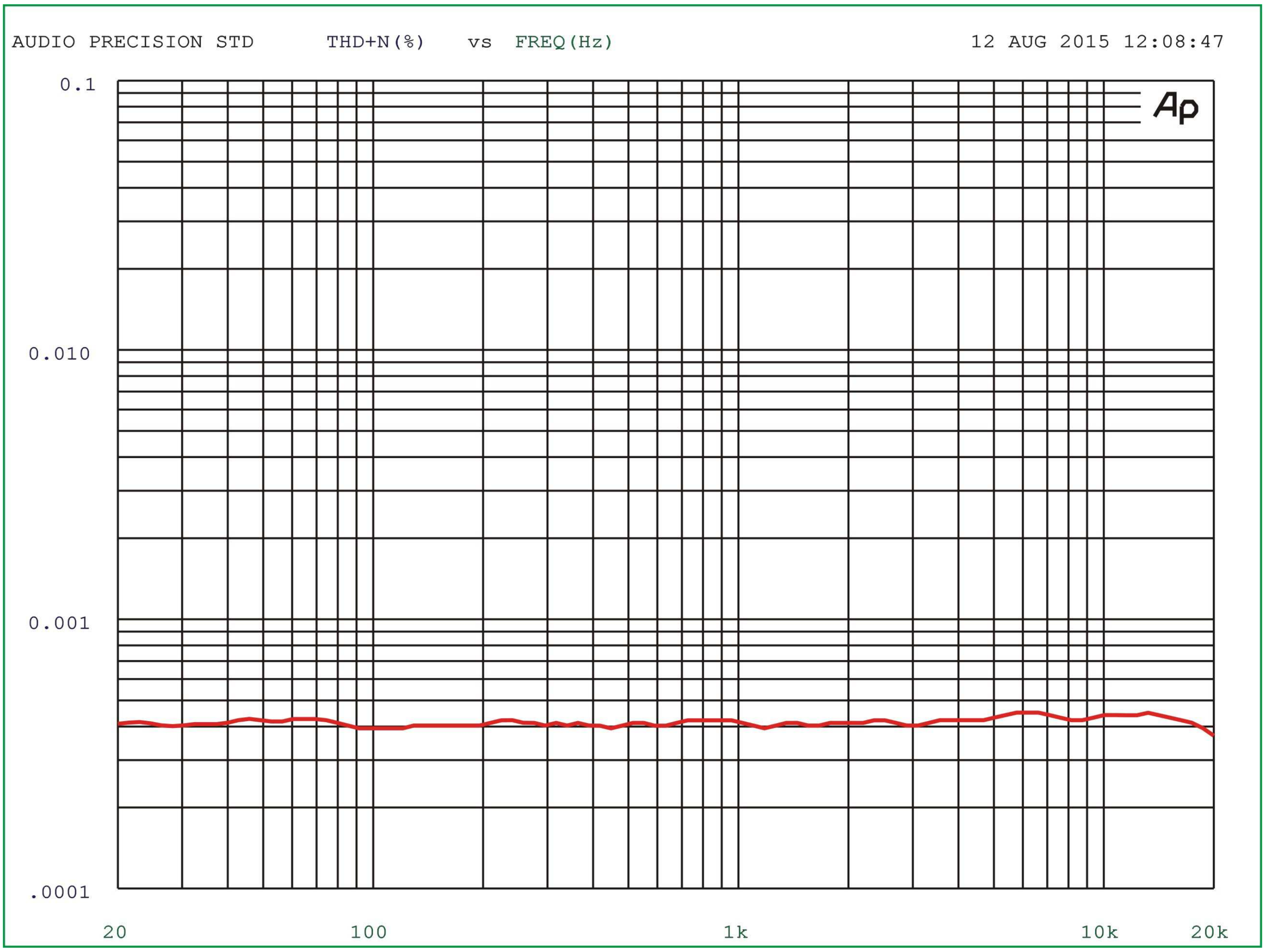
Distortion (THD+N) 20 Hz - 20 kHz @ -1.0 dBFS, sampling frequency 48 kHz



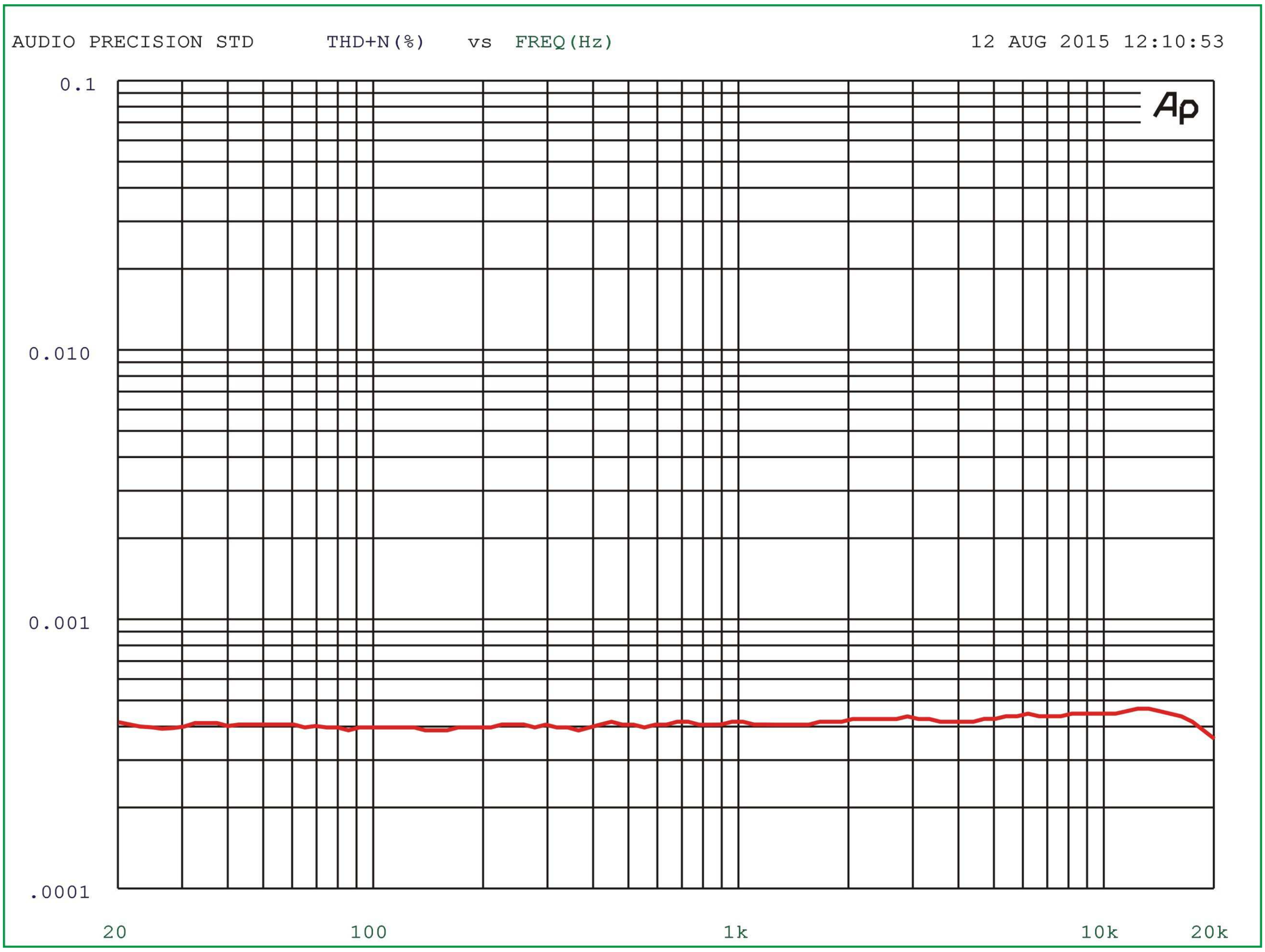
Distortion (THD+N) 20 Hz - 20 kHz @ -1.0 dBFS, sampling frequency 88.2 kHz



Distortion (THD+N) 20 Hz - 20 kHz @ -1.0 dBFS, sampling frequency 96 kHz



Distortion (THD+N) 20 Hz - 20 kHz @ -1.0 dBFS, sampling frequency 176.4 kHz



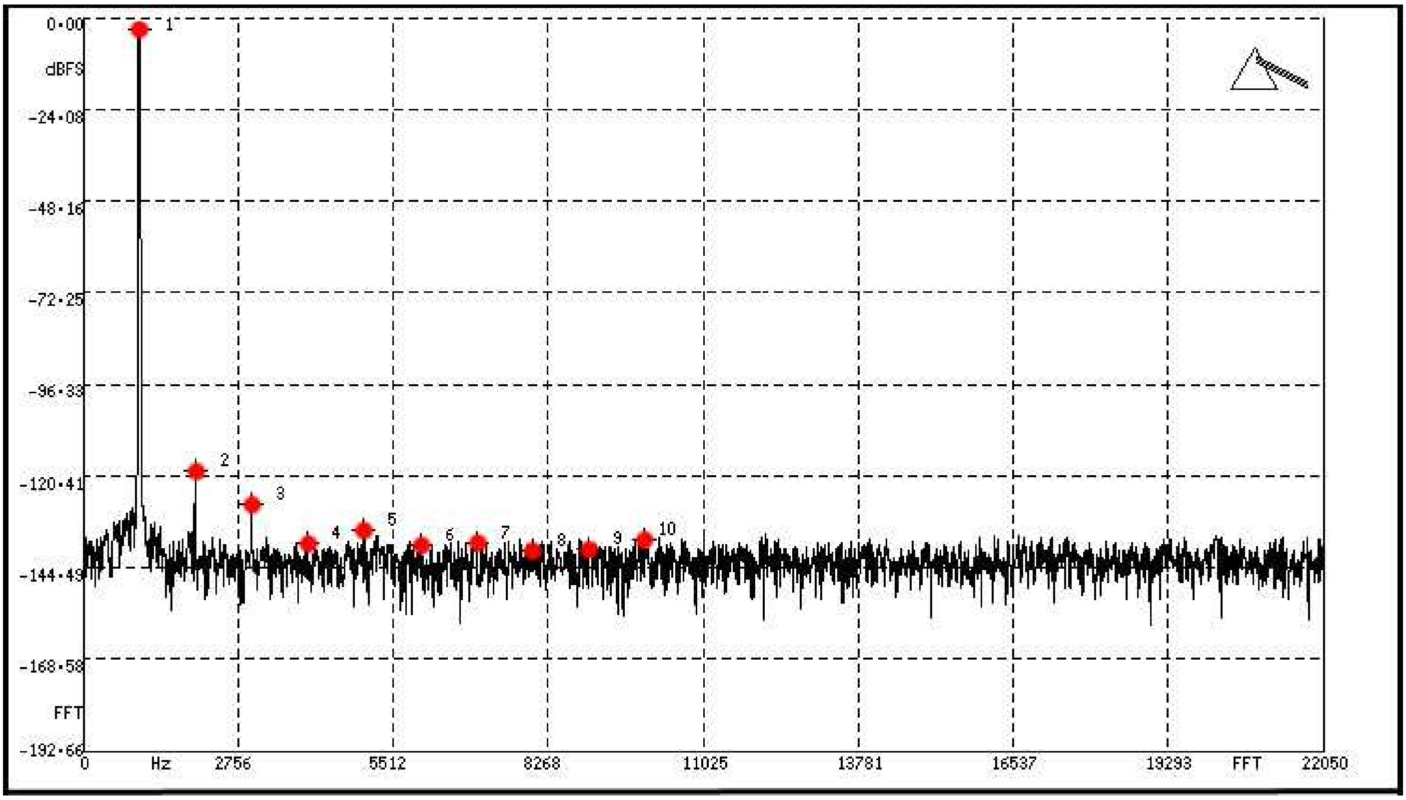
Distortion (THD+N) 20 Hz - 20 kHz @ -1.0 dBFS, sampling frequency 192 kHz

Note that the distortion+noise is less than 0.0005 % in the whole audio range, and virtually independent of sampling frequency.
Also, the distortion is largely linear with frequency, indicating that there are no slew rate/high frequency problems, and that sample rate and anti-aliasing filter coefficients are well balanced.

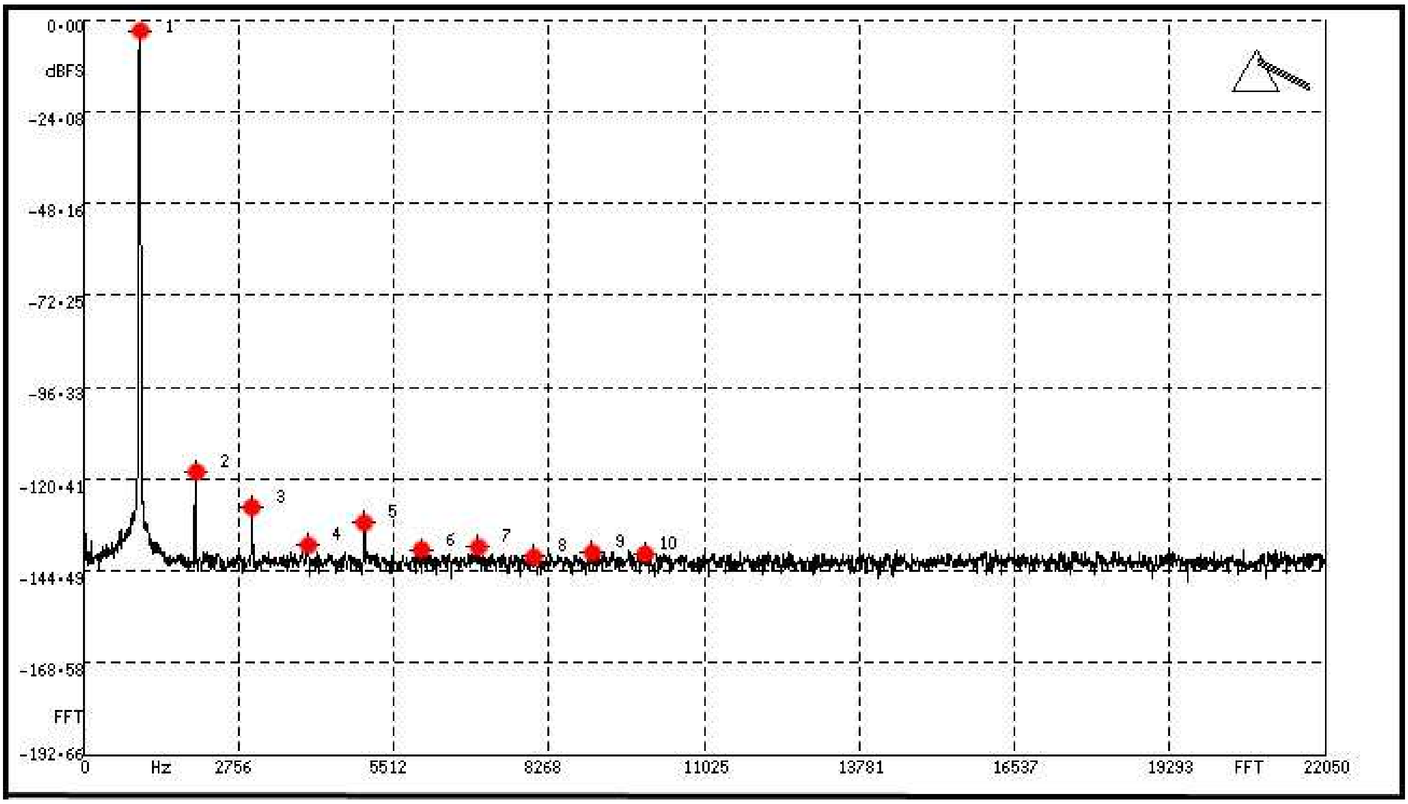
In depth analysis of distortion using FFT analysis

A more detailed picture of the distortion characteristics can be seen using FFT analysis. FFT analysis allows the individual distortion components to be quantified with a much greater accuracy than the traditional THD+N analysis can provide.

In the following FFT analysis only the distortion analysis at 44.1 kHz sample rate is included. Distortion characteristic at all the other sample rates are similar with only negligible variations in data.



This is a standard FFT analysis of the AD converter. The input level is -1 dBFS. Note that the majority of the higher harmonics are buried in the noise floor at -144 dB.



To get a clearer picture, the FFT was run 16 times with the average result calculated and displayed. The noise is partially cancelled out, because noise is a statistic phenomenon represented by random numbers. The distortion products are real constant numbers and are displayed at fixed positions.

The measured distortion result is impressive: The total RMS sum of all the THD components is -116.48 dBFS which equals 0.00015 %

And here it is in detail:

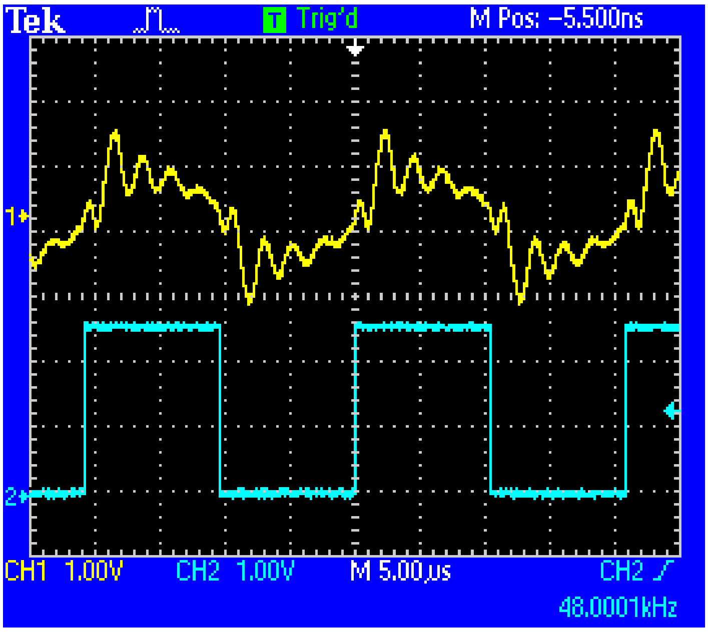
Marker	frequency	level dBFS
1. Fundamental	1 kHz	-1.0
2. Second harmonic	2 kHz	-118.28
3. Third harmonic	3 kHz	-127.38
4. Fourth harmonic	4 kHz	-136.94
5. Fifth harmonic	5 kHz	-131.45
6. Sixth harmonic	6 kHz	-138.70
7. Seventh harmonic	7 kHz	-137.78
8. Eight harmonic	8 kHz	-140.08
9. Ninth harmonic	9 kHz	-139.13
10. Tenth harmonic	10 kHz	-139.56

Synchronization and sync clean-up

One of the reasons for the excellent sonic characteristics of the AES module is the ultra low jitter clocks generated by the on-board clock generator. Providing a high quality clock, the generator supplies all the necessary system clocks internally to run the converters and the AES encoders.
AD converters are generally highly sensitive to clock jitter, and the excellent jitter characteristic of the generator is reflected in both the electrical specifications and in the perceived sound quality.

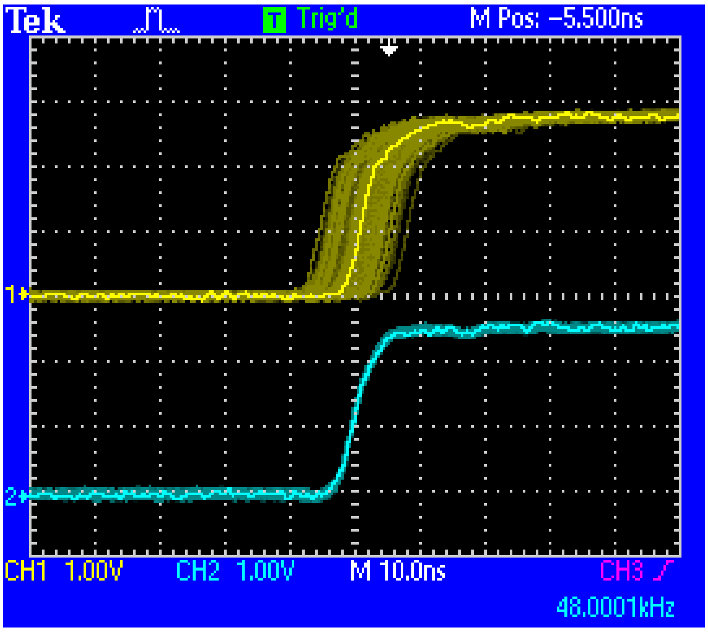
The AES module has to be able to synchronize to an incoming clock in the real world. Often this clock is of questionable quality, so in order to safeguard the high sound quality, it is necessary for the generator to have a comprehensive jitter and noise rejecting ability.
The on-board generator certainly has that. It is able to receive a totally smashed-up word clock and regenerate it into a pure high quality master clock for the circuits in the module. The scope snapshot below shows this ability. The upper trace is an incoming clock that is noisy and severely distorted. The lower trace shows the perfectly regenerated output clock.

The input uses a high-speed comparator with hysteresis and a "sweet spot" detector, which performs a highly 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 output clock.
This kind of distortion is usually due to low quality cable, wrong signal routing or wrong termination. Such a distortion is normally "steady state" which means that a repetitive and useable part of the signal exists. And if it exists, *the sweet spot detector will find it!*



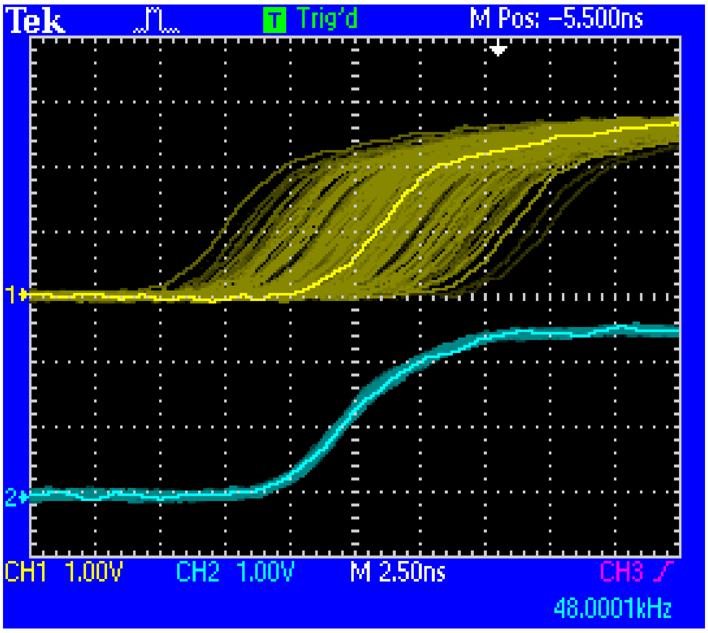
The scope pictures below show the excellent jitter rejection ability of the on-board generator.

In order to measure the jitter performance of the clock module, a highly stringent and revealing method was used. For the technical minded, the method is described in details in the ADDA16 data sheet; downloadable from the Rostec website.

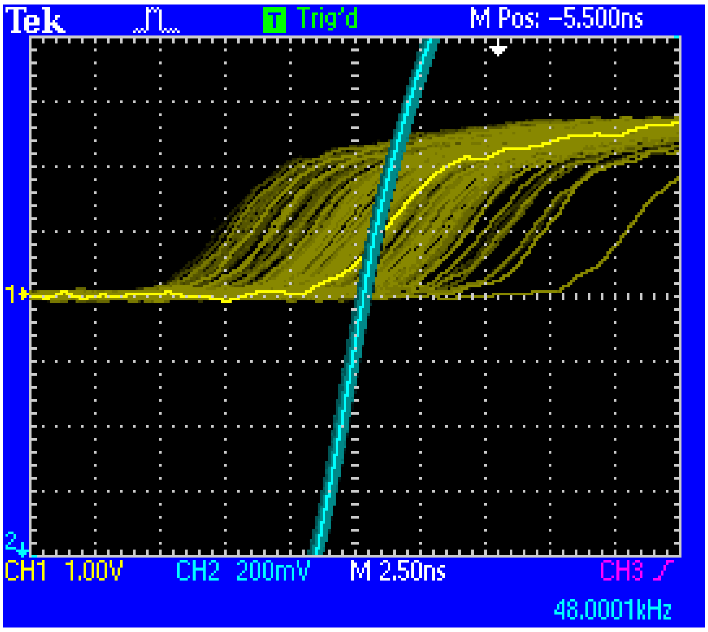


The upper trace shows the jittery clock sent to the input of the clock module and the lower trace shows the regenerated internal word clock with a nice and clean leading edge. The time scale is 10 nsec per division.

But let's take a closer look at 2.5 nsec per division



Let's look even closer



Adding 5x gain on the oscilloscope trace that shows the output jitter, reveals the true amount of jitter to be approx. 500 psec PP, which is approx 80 psec RMS. (0.08 nsec)

It does not matter whether external sync lock or internal crystal reference is used. The result is the same. The jitter performance and the output clock quality are always invariably excellent.

Note: These are the internal word clock specs with an extremely jittery input clock. All the other system clocks (master clock, serial clock, data clock etc.) have jitter specs in the order of 2 – 8 psec. If you want to measure these accurately, you may have to ask NASA for help.