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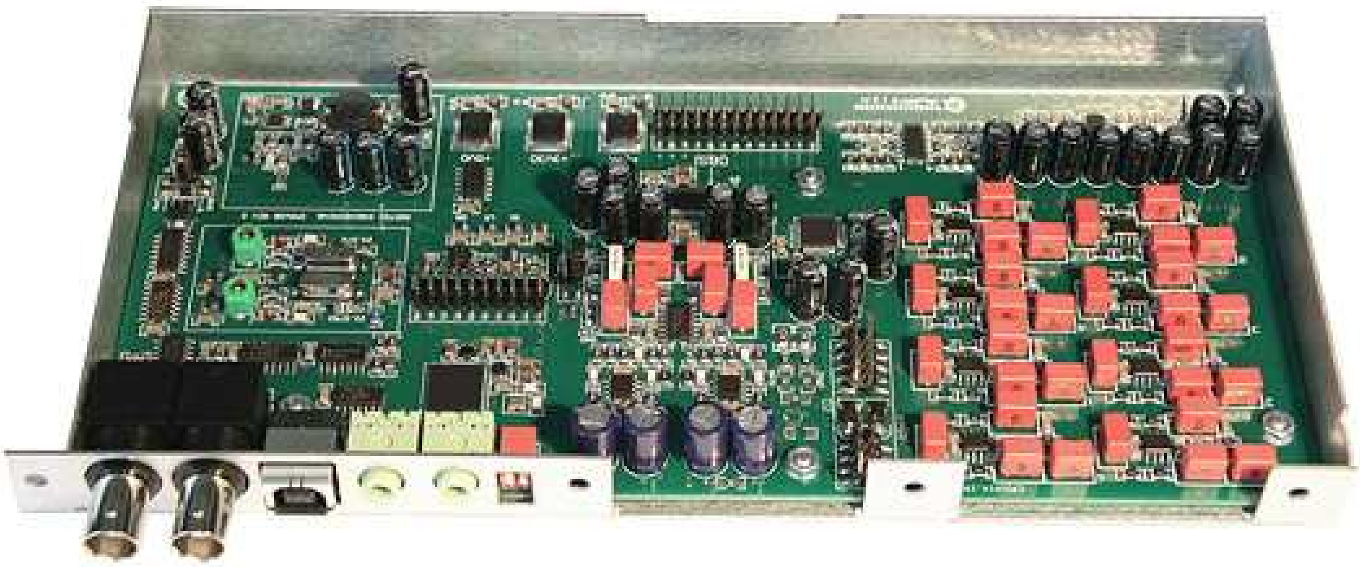

USB 2.0 Module
for ROSTEC LMA8 preamplifier

Revision 3, February 10-2014

USB 2.0 Module for LMA8 Mic/Line Preamplifier

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Features

- Fully compliant USB 2.0 Audio Interface
- Bit-perfect USB data transfer
- 8 channels of 192 kHz, 24-bit, 118 dB ADC
- 2 channels of 192 kHz, 24-bit, 120 dB DAC
- Large headroom at all point in the signal chain
- Impressive low distortion specifications
- Fully balanced analog architecture for immunity against interference from ground noise, digital clocks, power lines etc
- Exceptional low jitter on-board clock generator
- Industrial output buffers with +30 dBu headroom
- Combination of JFET and BIPOLAR technology for exquisite sonic performance
- Taps the audio signal directly at the preamp circuit for shortest possible signal path.
- Word sync input has a "sweet spot" detector with signal clean-up and clock regeneration
- Works brilliantly as a high performance Master Word Clock Generator for the whole studio
- No power drain from the USB connection
- Works directly without drivers on Mac systems, Windows systems need a driver (pending)
- Inputs and outputs are ESD protected to 23 kV, IEC 61000-4-2 and 15 A surges, IEC 61000-4-5.

General description

The USB 2.0 Module is a fully self-contained 8 input/2 output digital interface for the LMA8 analog preamplifier. It features 8 channels of AD and 2 channels of DA using high performance 192 kHz 24 bit converters.
The analog circuits are state of the art using high quality top range components, and all analog filters are of minimum phase design with well controlled impulse response
The USB interface processor provides bit-perfect USB audio streaming to and from the DAW. When plugging in the USB connector on a Mac system, the 8 input and 2 output channels will immediately be available in the operating system and in the recording software.

The module also features a high performance clock generator with word clock sync capability.
The clock generator is able to synchronize to all the standard word clock frequencies, and it performs an efficient signal clean-up of the incoming clock, resulting in a perfect output clock (se technical section).
When no sync is present at the input, the generator works smoothly and elegantly as a high quality master clock generator for the whole recording studio.
The sample rate can be selected from inside the operating system or the recording software.

Installation

The module is installed in 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 the module is installed, only the connector panel is accessible to the user, as shown on the picture below.

Analog inputs

The analog inputs of the module are not directly accessible to the user. The audio signal inputs are routed from the motherboard of the LMA8 via a multi-pin connector
The signal is tapped directly at the output of the preamp as a balanced signal and fed to the balanced AD pre-buffers.

Digital output (send to DAW)

The 8 digital audio signals from the AD converters are converted to a serial audio stream, and sent via the USB link to the DAW, where they immediately become available in the operating system and in the recording software.

Digital input (return from DAW)

The USB link also functions as a return path for the 2 output channels. The serial audio stream, received from the DAW, is converted to standard format and sent to the DA converter for conversion to analog signals.

Analog outputs

The balanced analog signal from the DA is sent to the 2 floating industrial output buffers providing the balanced audio outputs, available at the green mini jacks at the connector panel.
The tip of the jack connector is the positive signal, the ring is the negative signal and the body is the ground.

If an unbalanced connection is used, the ring (negative) <u>must</u> be shorted to ground.

OBS: Do no be fooled by the small inconspicuous jack connectors! They are internally gold plated, and behind them are two very capable output buffers with excellent technical specs and extraordinary sonic qualities.

About overload, clipping, headroom etc

The USB module works in perfect harmony with the LMA8 basic analog version 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.

At digital full scale, the analog level of the outputs at the connector panel of the USB module is +20 dBu
(can be switched to +4 dBu).

All output buffers have a max output level of +30 dBu, which means that the buffers always have a 10 dB of extra headroom at digital full scale!
<u>Analog output clipping is simply not an issue,</u>
The buffers always operate comfortably within their dynamic range!

When the signal level from the LMA8 mic preamp is high enough to bring about digital clipping in the AD chain, the mic preamp still has 16 dB of extra headroom!
So <u>input clipping of the mic preamp is also not an issue</u> because all other circuits (buffers, filters, ADs) have gone into clipping long before the preamp clips.

The advantage of this "comfy-zone" philosophy is very rewarding.
With the extraordinary large headroom at all points of the signal chain, the LMA8 and the USB module <u>always operate well controlled and firmly within their respective linear ranges.</u>

The benefit is evident. The LMA8, with the USB 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, so digital clipping is totally regular and well controlled. The protection circuit is active just above the clipping point, and does NOT interfere with the audio signal in any way!

"Actually, the clipping sounds kinda 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 Word Clock Generator controls all the digital clocks used internally by the USB processor and the AD/DA converters. At the same time, it provides a word clock output at the BNC connector at the connector panel of the USB module.
The sampling frequency is selected from within the operating system or from within the recording software.

The generator detects when an incoming word clock is present at the BNC connector at the connector panel, and if the incoming clock is of the correct frequency, the generator locks on to it immediately. It softly glides from the internal crystal reference to the external clock reference without any jumps or disruptions of the clock signal.
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 never any gaps or interruption of the word clock signal or the internal clocks.

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 USB 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 an output clock with typically less than 0.1 nsec RMS jitter.

This exceptional jitter performance is one of the reasons for the exceptional sonic quality of the USB module

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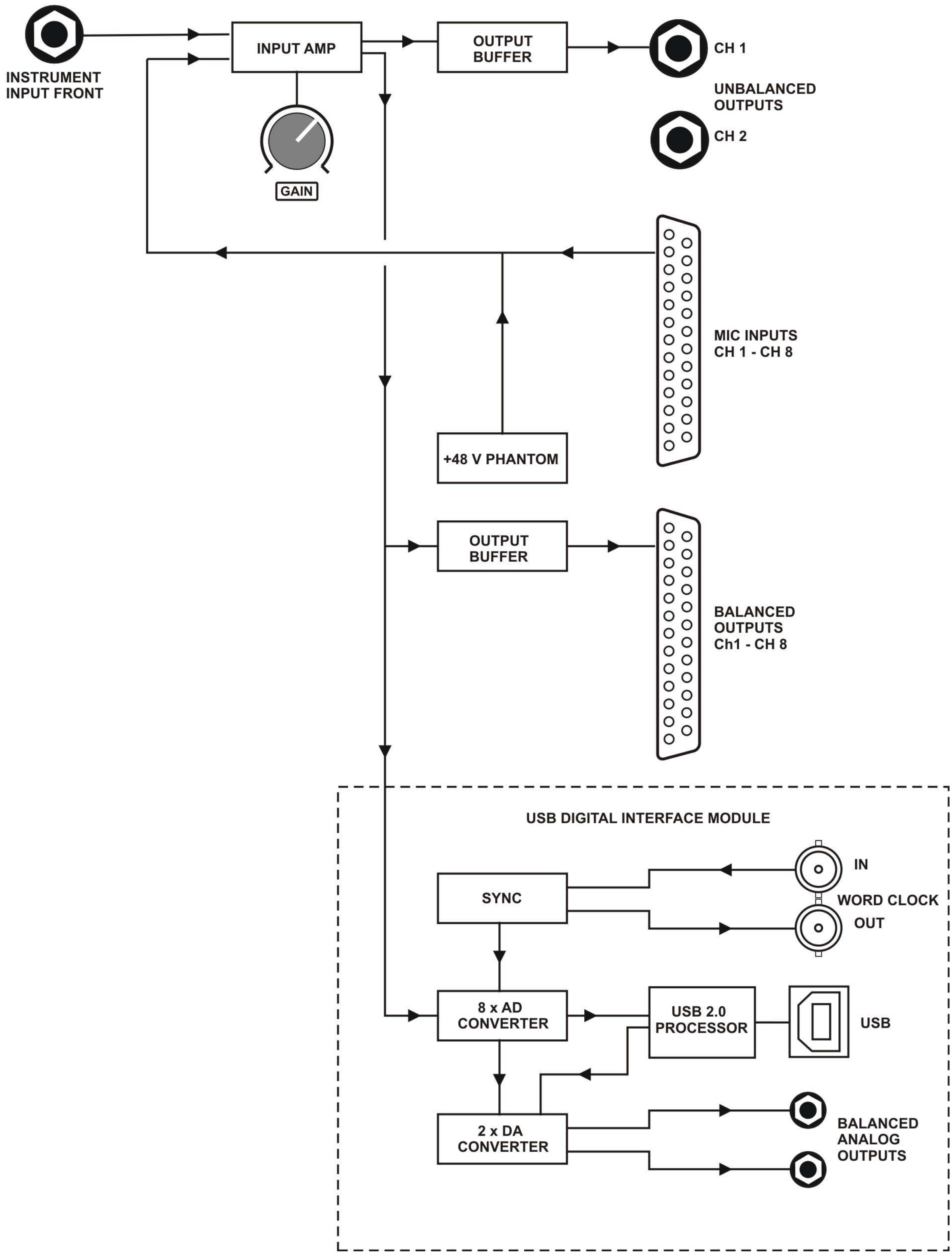
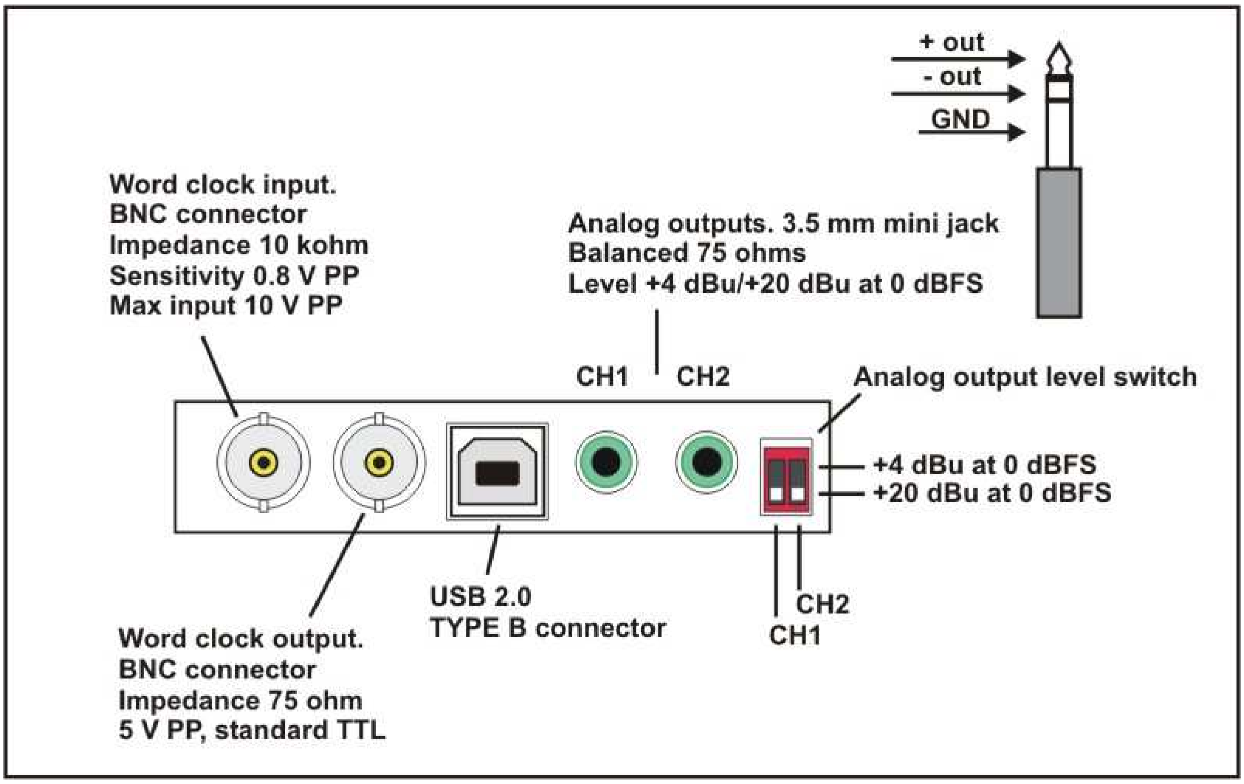
LMA8 with USB module, simplified signal flow, 1 channel shown



Connector panel quick guide



Mechanical and electrical specifications

Dimensions: Width 230 mm, height 20 mm, depth 120 mm
Weight: 1.0 kg
Power requirements: +24 V, +15 V, -15 V, 6.5 Watts
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: 2 ps RMS
Word clock output jitter: 100 ps RMS
PLL capture range: +/-50 ppm
Lock time approx: 0.2 - 0.4 sec

Word Clock input impedance:	2.2 kohm
Word Clock input level:	Min 0.8 V PP, Max 10 V PP
Word Clock output impedance:	75 ohms
Word Clock output level:	5 V PP, TTL Level

USB:
USB Audio Class 2.0
Direct interface to Mac systems, OS X 10.6.8 or later.
Windows not supported (pending)

AD converter:
8 channels

Resolution:	Max 192 kHz 24 bit
Dynamic range:	118 dB, limited to 116 dB by buffer noise
THD+N:	0.0004 %
THD:	0.00018 %

DA converter:
2 channels

Resolution:	Max 192 kHz 24 bit
Dynamic range	120 dB, limited to 116 dB by buffer noise
THD+N:	0.0006 %
THD:	0.0003 %

Analog:
Analog input level: Internally defined (no user access)

Analog output level:	+20 dBu/+4 dBu at 0 dBFS
Crosstalk adjacent channel:	-126 dB at 1 kHz

Technical section

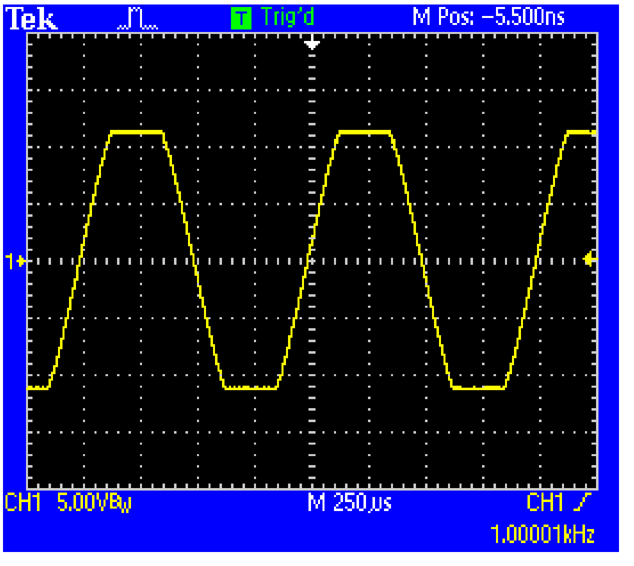
Clipping characteristics

Digital clipping has always had a bad name, and with good reason. Many AD converters sound like a total system crash when they are forced into clipping. Various manufacturers have employed elaborate solutions to compensate for this, like input compressors and "soft limiters" etc. In reality, it is just swapping one bad thing for another.
Rostec uses a different approach. The USB 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 USB, and sent back from the DAW via USB to the 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.

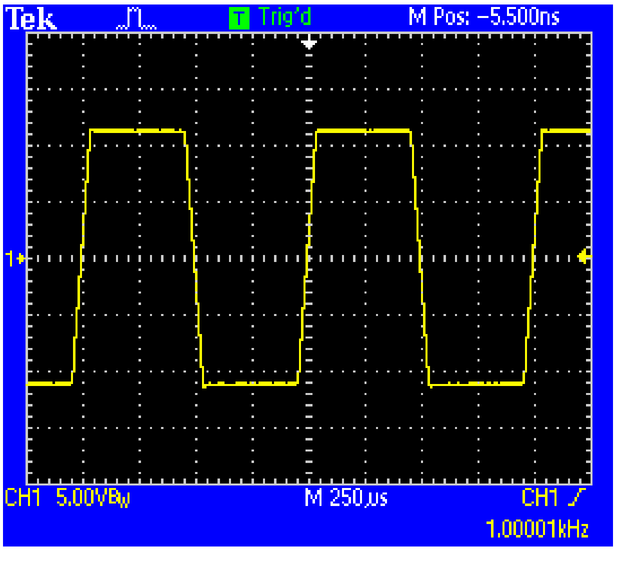
+3 dB overload
Sampling frequency: **192 kHz**
Analog frequency: 1 kHz
Measurement bandwidth: 200 MHz.

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.



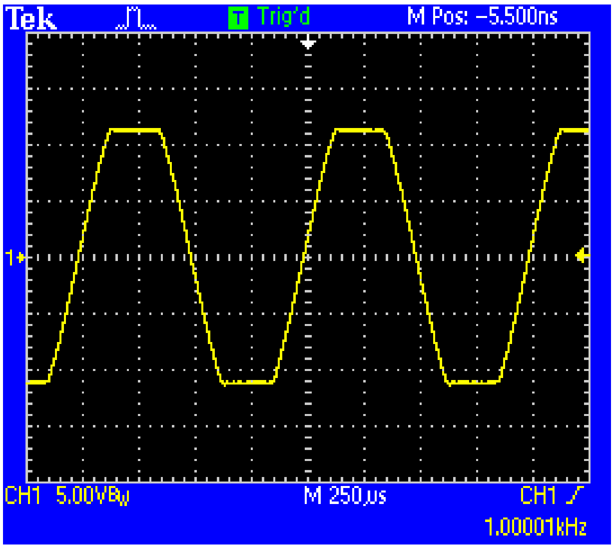
+12 dB overload
Sampling frequency: **192 kHz**
Analog frequency: 1 kHz
Measurement bandwidth: 200 MHz.

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.



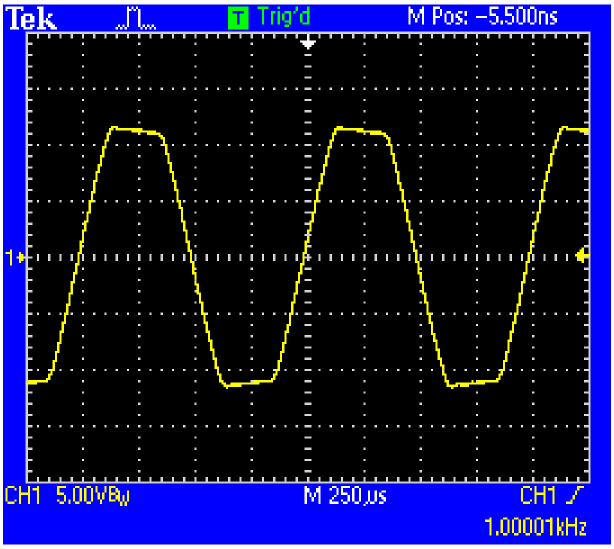
+3 dB overload
Sampling frequency**: 96 kHz**
Analog frequency: 1 kHz
Measurement bandwidth: 200 MHz.

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



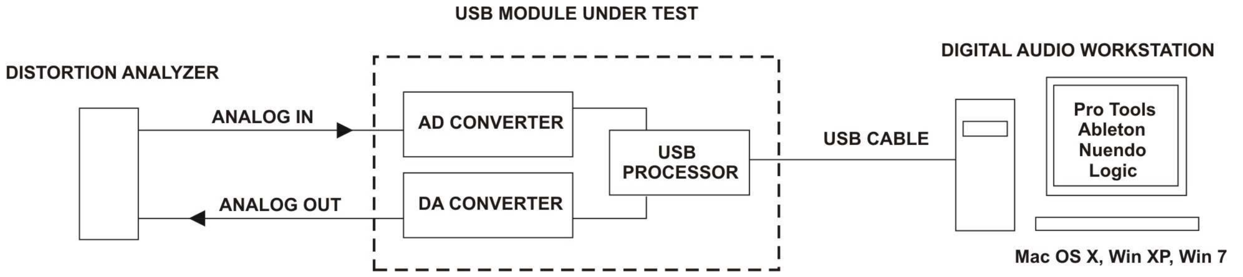
+3 dB overload
Sampling frequency**: 48 kHz**
Analog frequency: 1 kHz
Measurement bandwidth: 200 MHz.

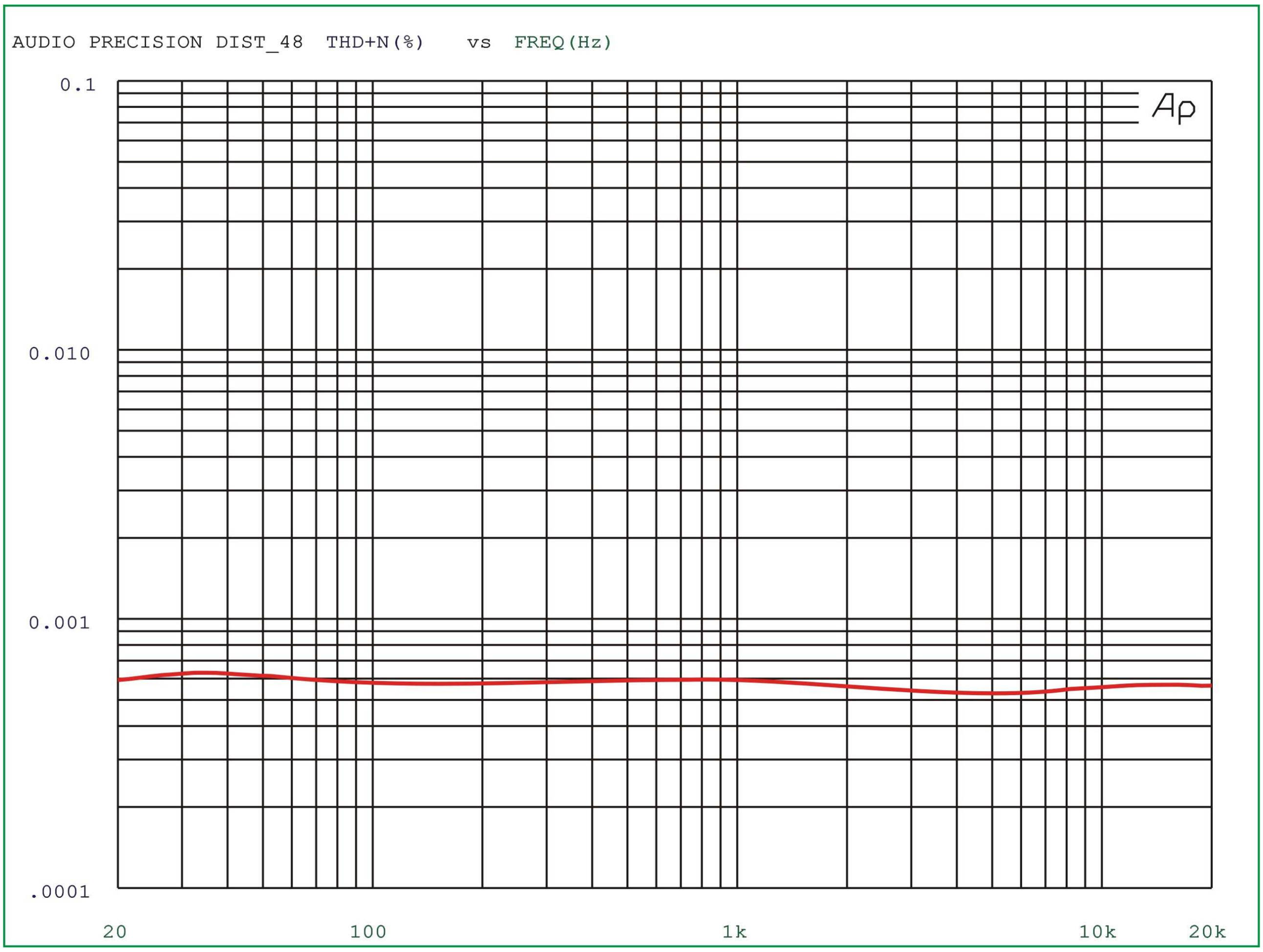
At 48 kHz sampling frequency, the clipping shows a less regular cut. At first glance it looks like lo-cut filtering, but it is actually due to a small ringing from the 100 dB digital anti-aliasing filter in the converter chip. There is absolutely nothing unusual in this. This is how digital filters work.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



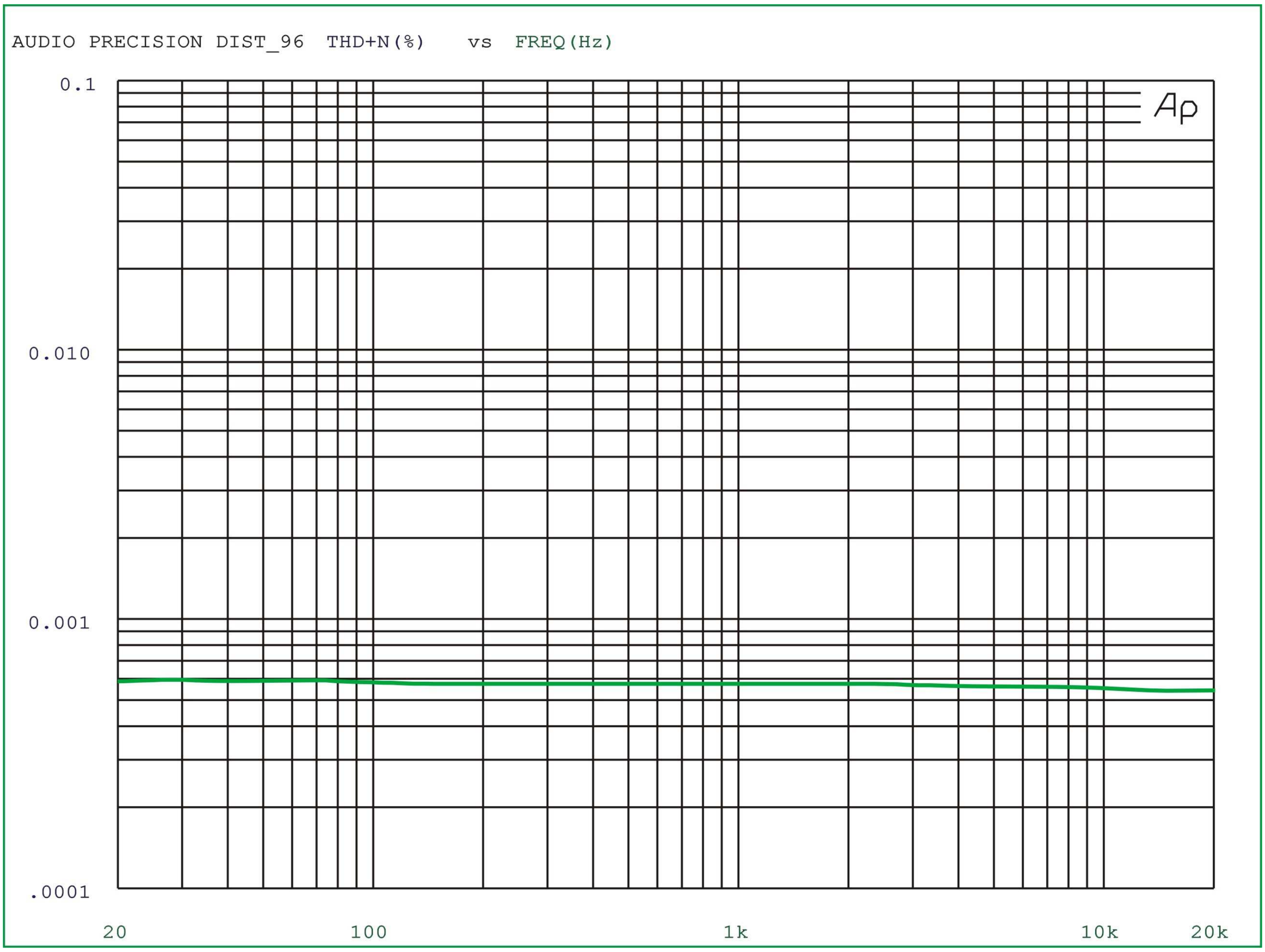
Distortion characteristics

The distortion test was performed as a "Real World Test". An analog signal was sent to the module; the signal was sent via USB to a DAW and returned via USB from the DAW. The distortion was then measured at the analog output of the module, as shown on the block schematic below.

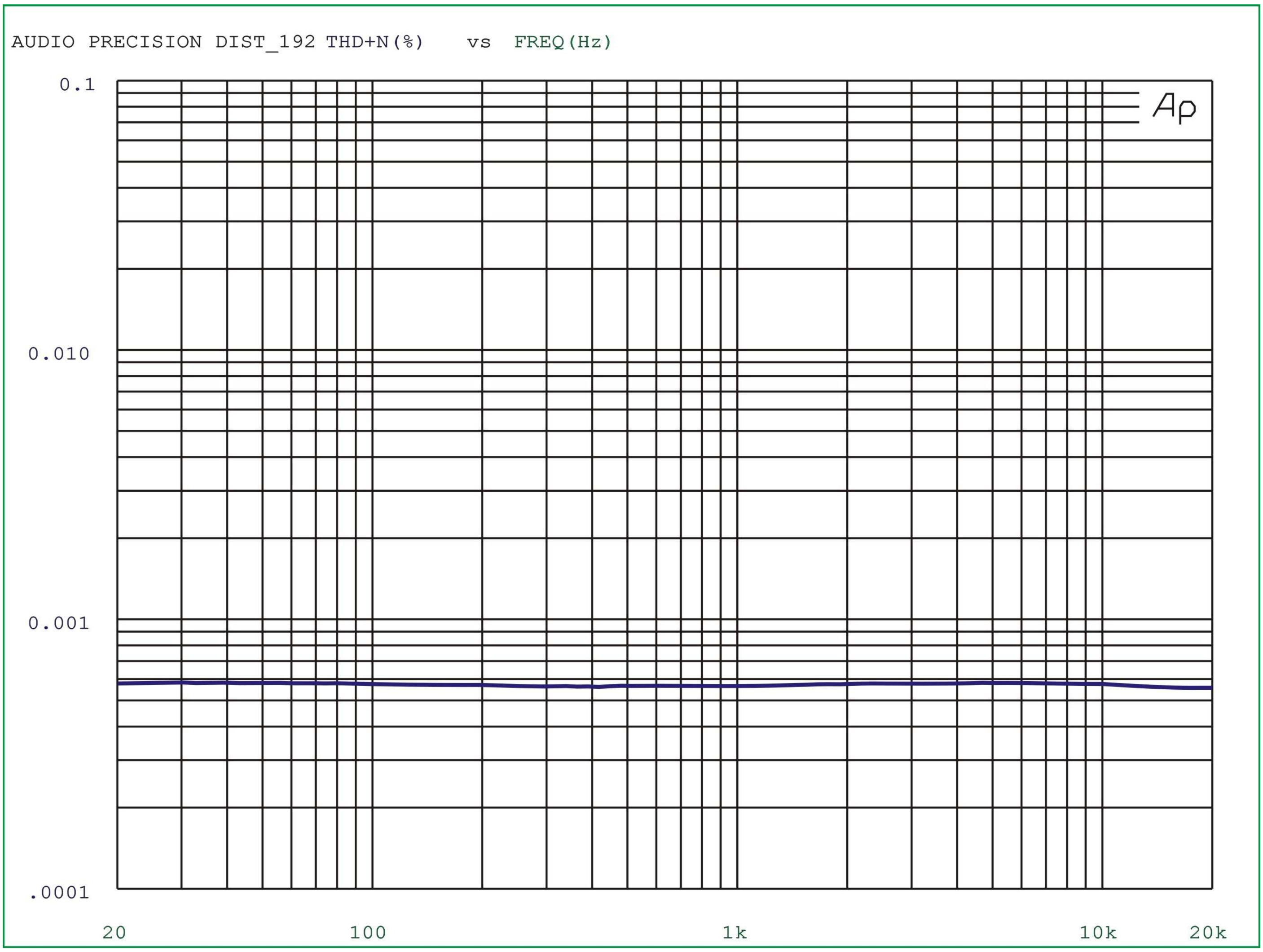




Distortion (THD+N) 20 Hz - 20 kHz @ -1.0 dBFS, sampling frequency 48 kHz



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



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

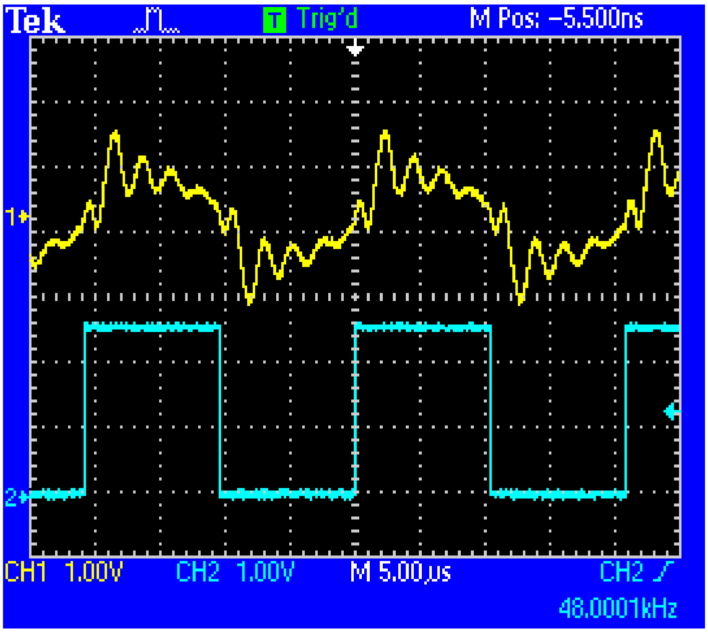
Note that the distortion is better than 0.0006 % in the whole audio range, and virtually independent of sampling frequency. The distortion is largely linear with frequency, indicating that there are no slew rate/high frequency problems, and that sample rate and anti-aliasing filter coefficients are well balanced.

Synchronization and sync clean-up

One of the reasons for the excellent sonic characteristics of the USB module is the ultra low jitter clocks generated by the on-board clock generator. Besides providing a high quality word clock output, the generator also supplies all the necessary clocks internally to run the converters and the USB processor.
All DA and AD converters are 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.

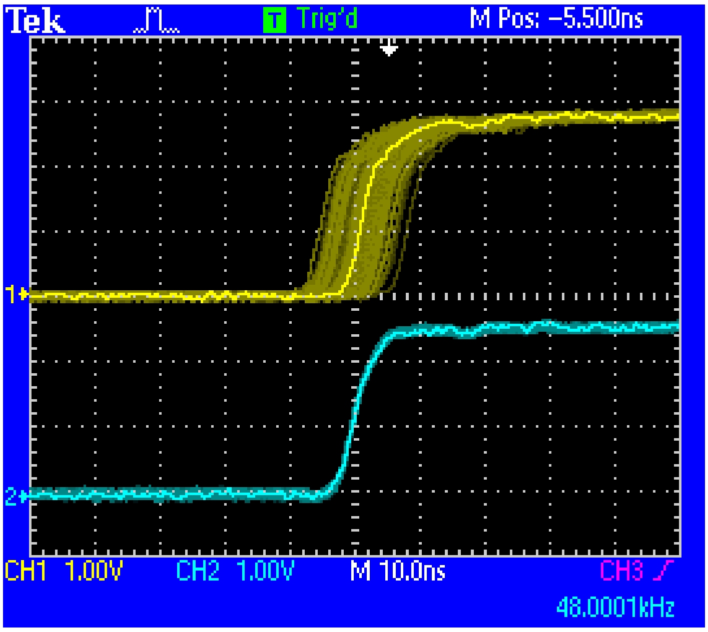
The USB module has to be able to synchronize to an incoming clock in the real world. Often this clock is of questionable quality, so in order to safeguard the high sound quality, it is necessary for the generator to have a comprehensive jitter and noise rejecting ability.
The on-board generator certainly has that. It is able to receive a totally smashed-up word clock and regenerate it into a pure high quality output word clock. 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!*



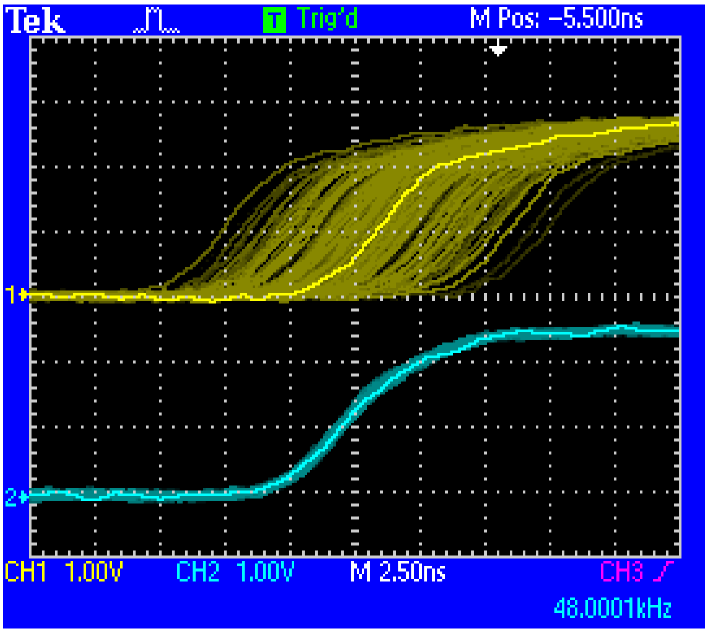
The scope pictures below show the excellent jitter rejection ability of the on-board generator.

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.



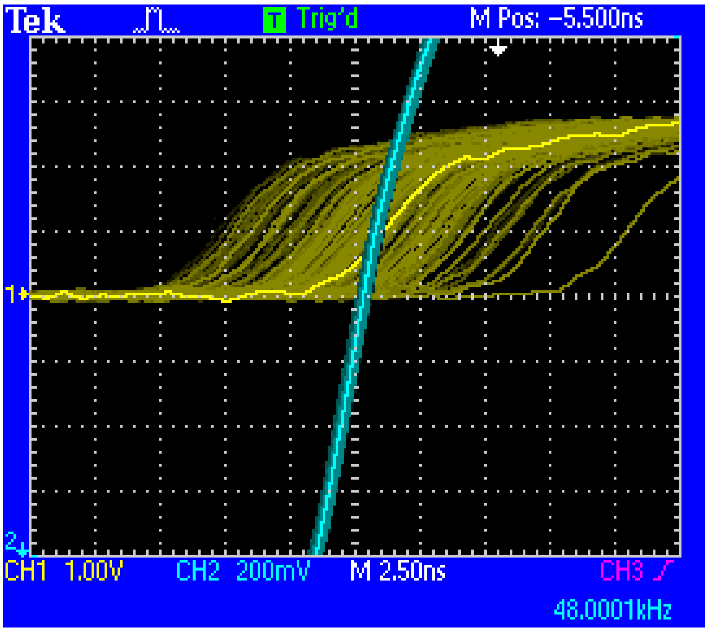
The upper trace shows the jittery clock sent to the input of the clock module and the lower trace shows the regenerated output clock with a nice and clean leading edge. The time scale is 10 nsec per division.

But let's take a closer look



At 2.5 nsec per division, the output jitter begins to become visible.

So let's look even closer



Adding 5x gain on the 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)

Whether external sync lock is used, or internal crystal reference is used, is immaterial. It simply doesn't matter. The jitter performance and the output clock quality are always invariably excellent.