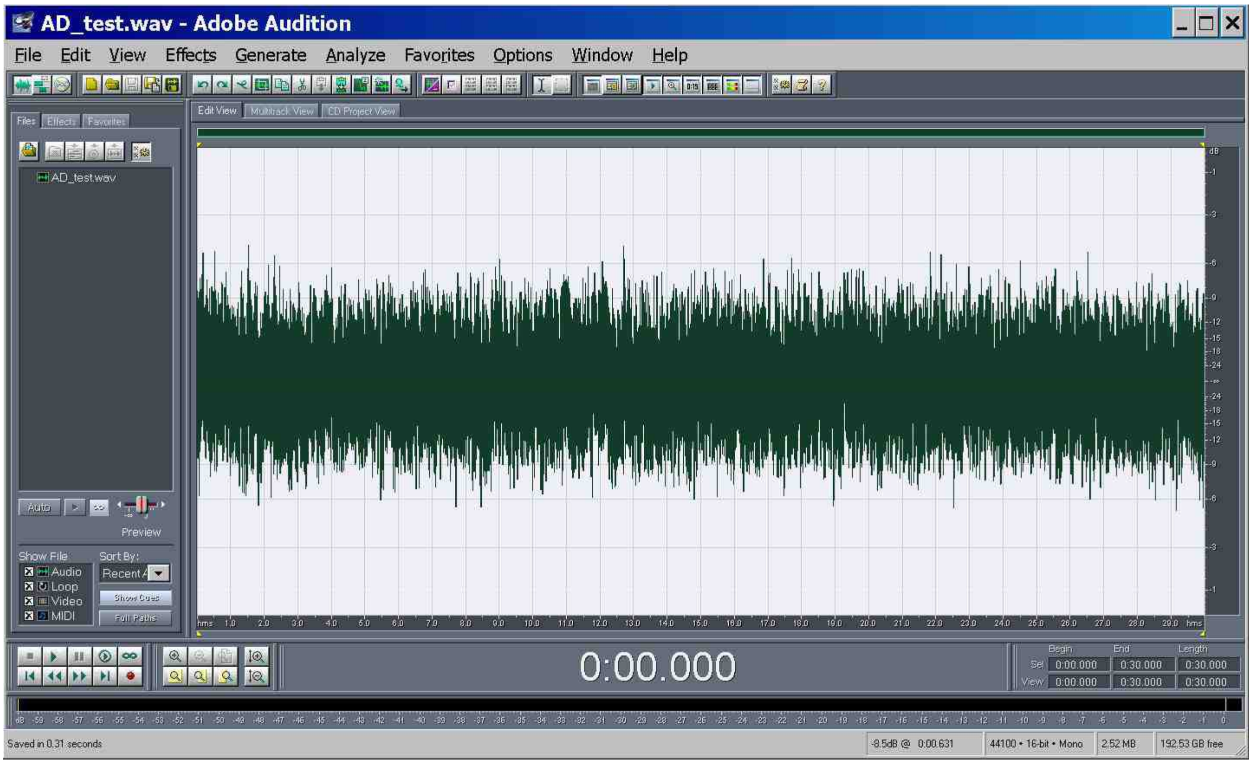
A simple method of estimation for audio equipment linearity and errors of digital audio interfaces

At the end of the 1997, I was mixing commercials and music on PC and installed a sound editor program "COOL Edit". After studying the rich editor abilities of the filtering and spectral analysis, I decided to use them to evaluate the linearity of the audio signal path. The idea was simple: to create an audio file of pink noise (it perfectly simulates music), trim some parts of the spectrum (make empty gaps), burn the file to a CD, play the disc and use a spectrum analyzer to estimate (via sound card) the output of the CD-player for spurious spectral components inevitably appeared within gaps due to a nonlinearity of analog channel and timing/handling errors of digital path.

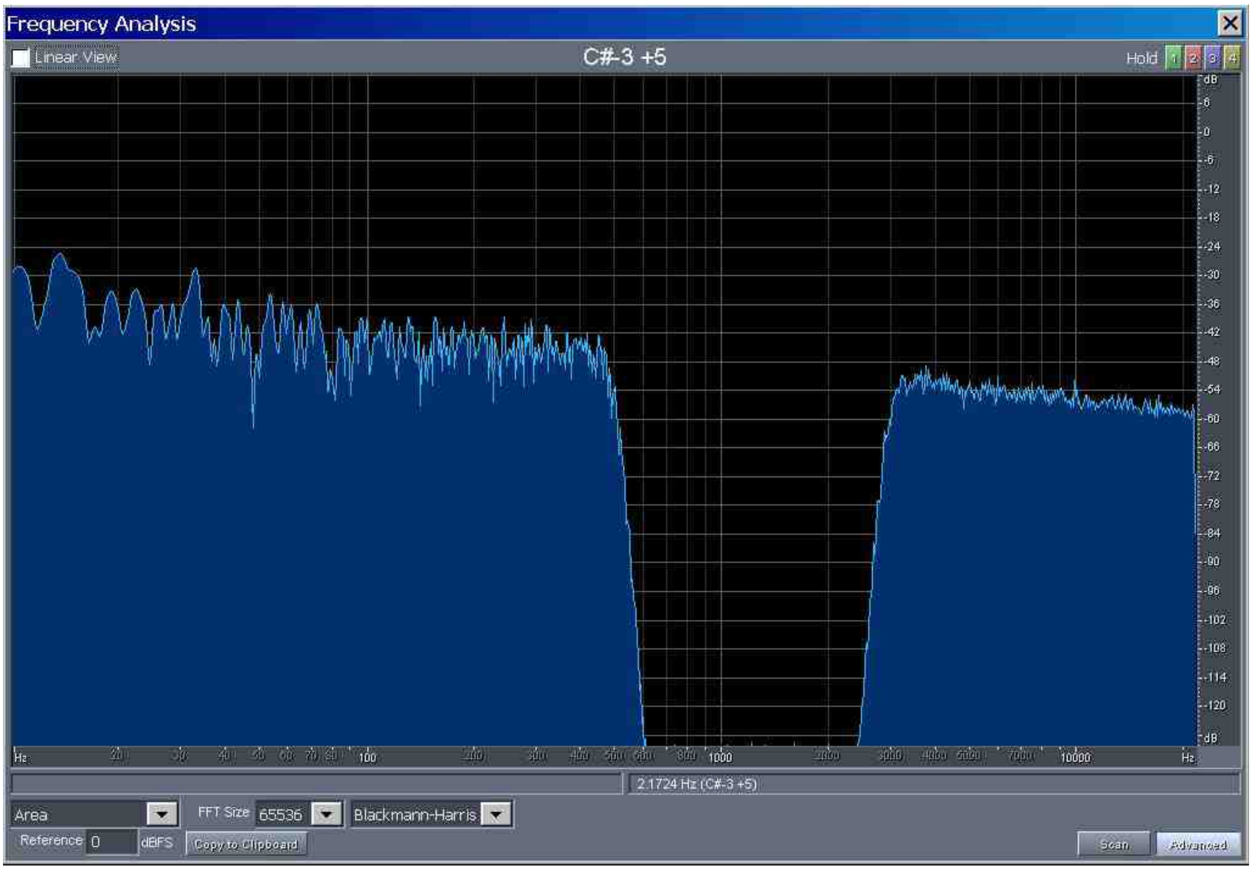
The technology was simple, and after a while, I saw in the editor window a "mud" that the CD-player introduced. Then I outputted the same file via S/PDIF to eliminate the player DAC. I was surprised even more: the peak level of new spectral lines reached -54 dB regardless of the optical or coaxial interface implementation. Examining this way some studio sound devices, I was glad the speed and simplicity of linearity estimation of *any* audio path from source to speaker.

In 2001, I began to prepare my book [1] for publication and when I was seeking references on chapter "Tests of amplifiers," I found Belcher's article [2], in which he described the Distortion measurement with p.r. noise and two comb filters. The method used Belcher was much more difficult to implement, since there were no personal computers with the required hardware and software at that date. However, I realized that I was on the right path.

Recently I updated my test file (see below) and made with it several interesting measurements.

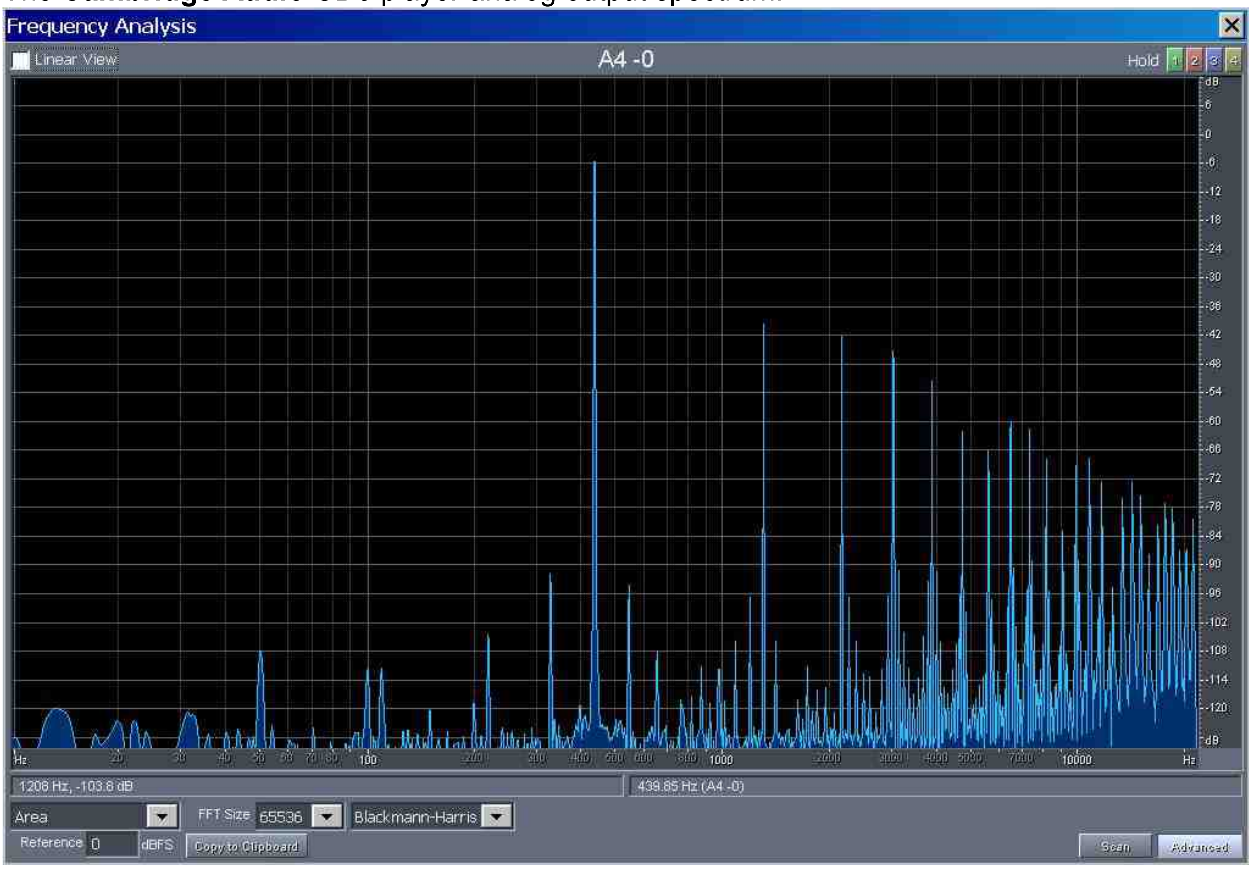


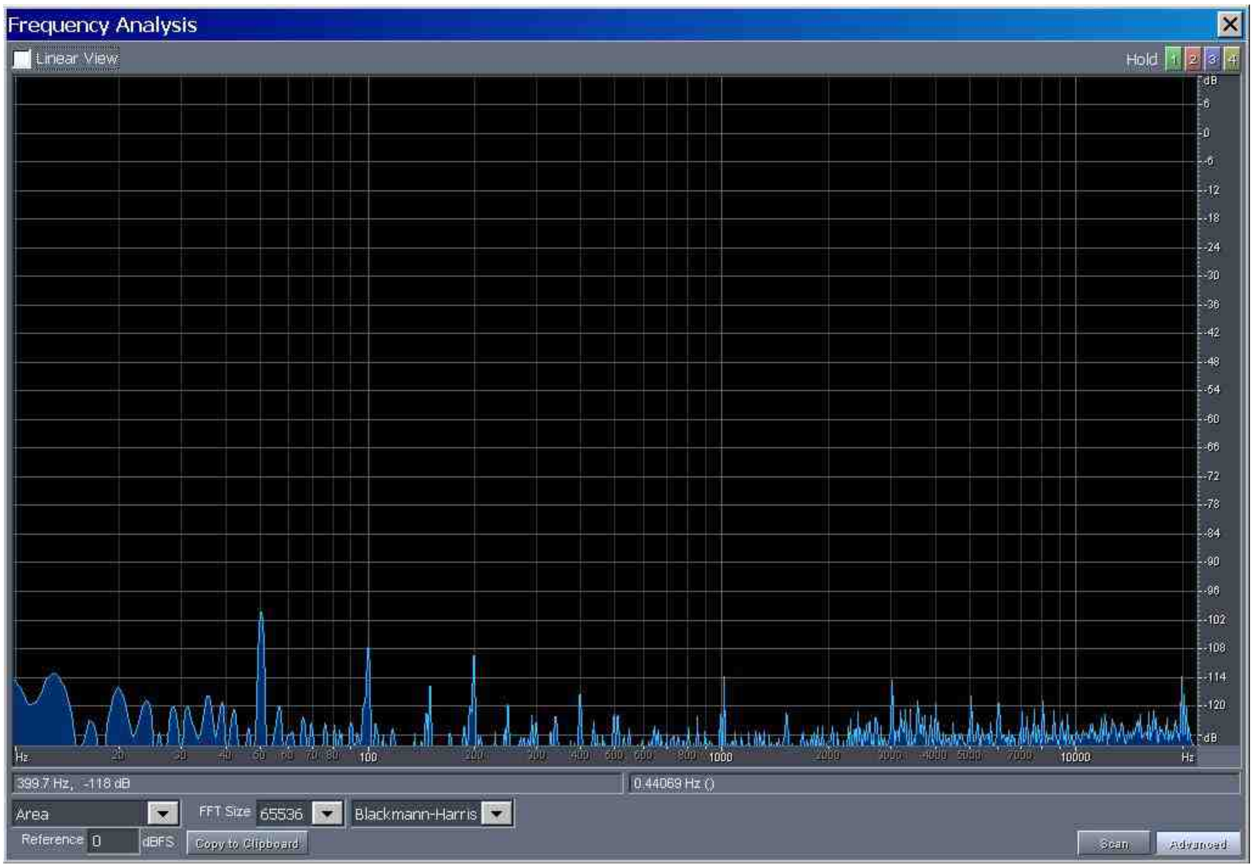
And its spectrum

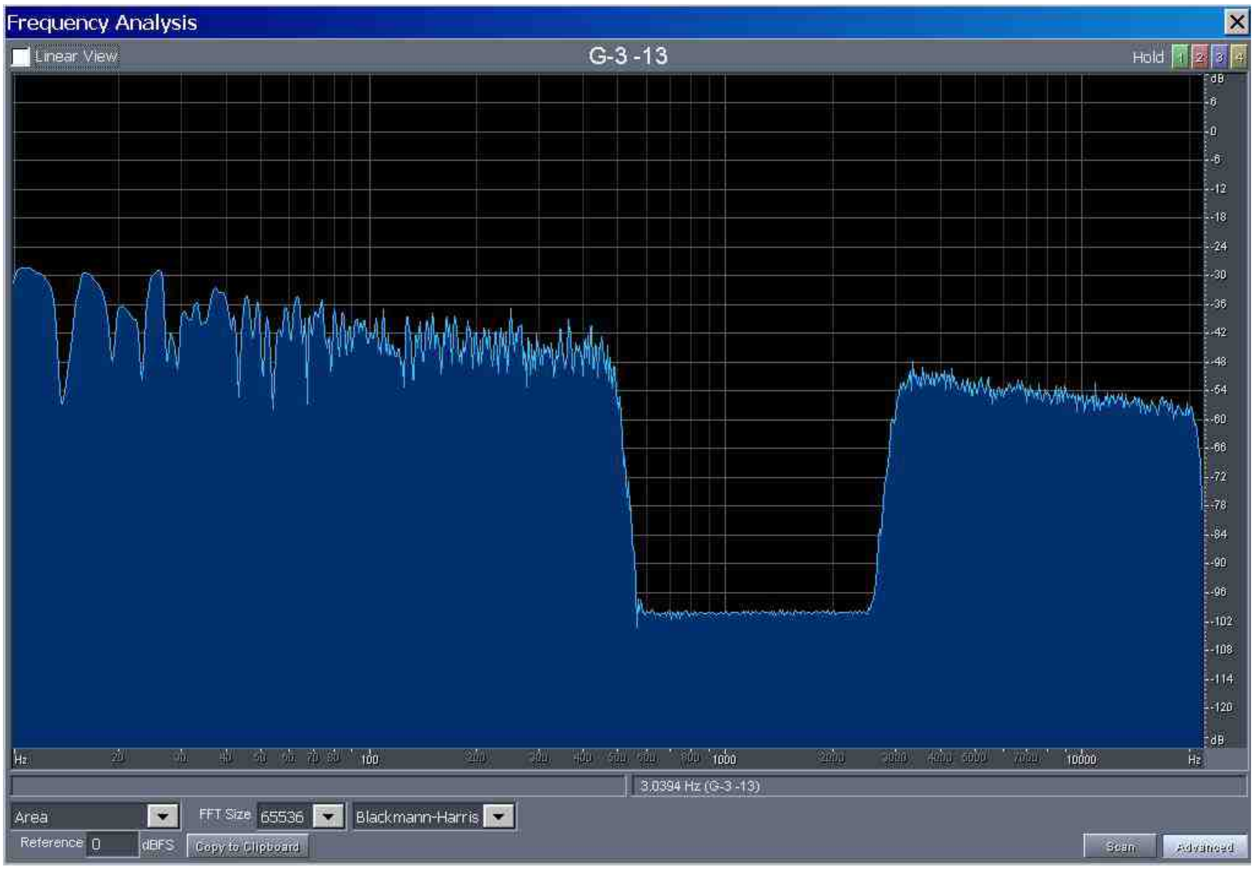


I recorded the test signal onto CD with two of reference tracks: –6 dB 440 Hz signal and silence signal. Measured signals shown below.

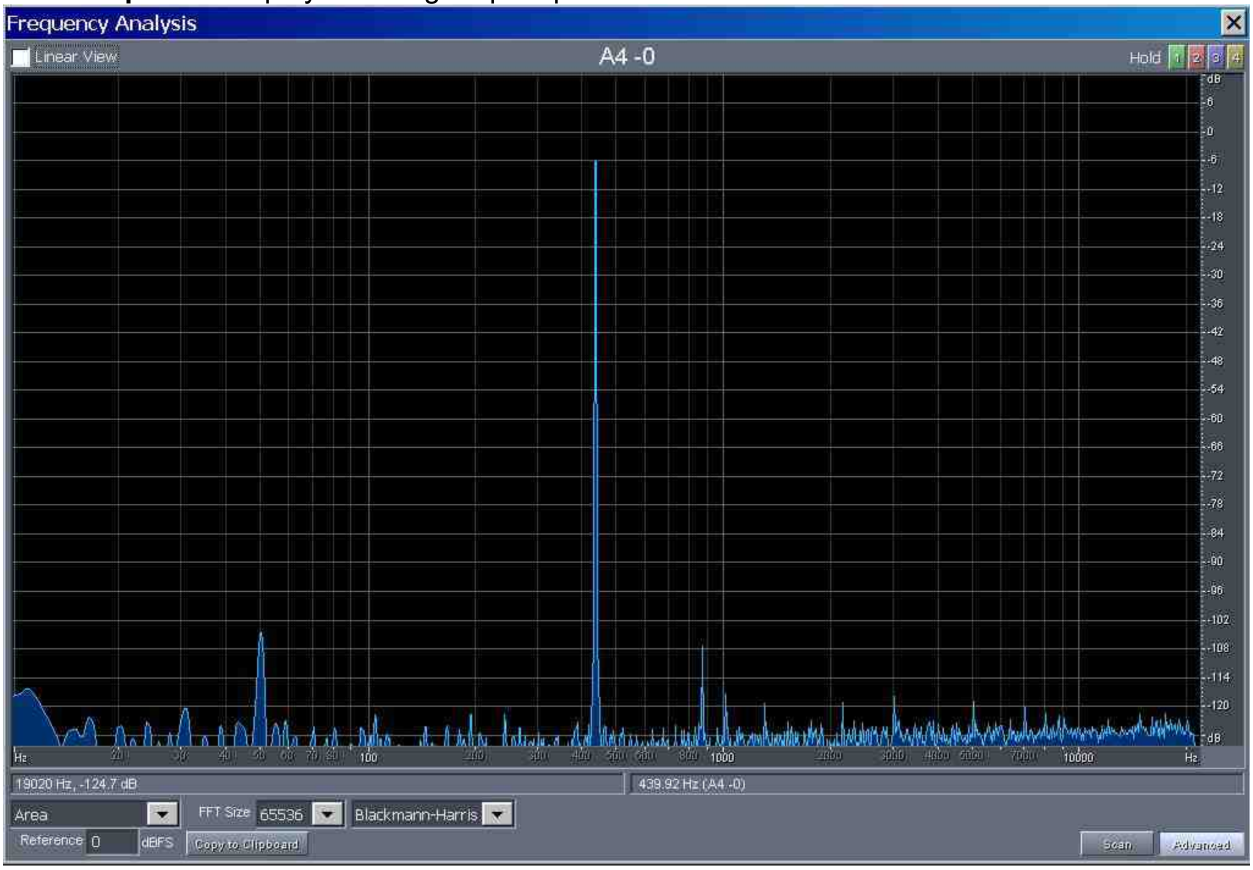
The **Cambridge Audio CD6** player analog output spectrum:

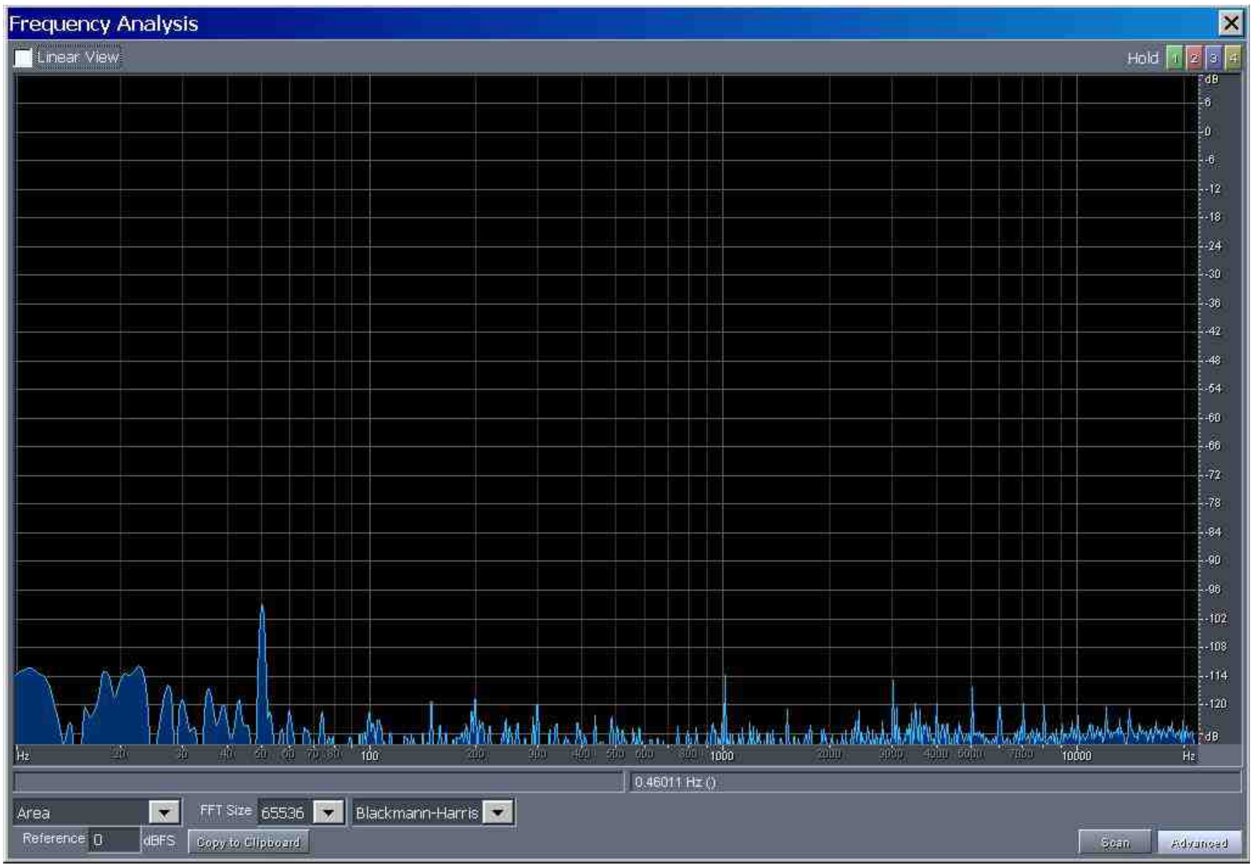


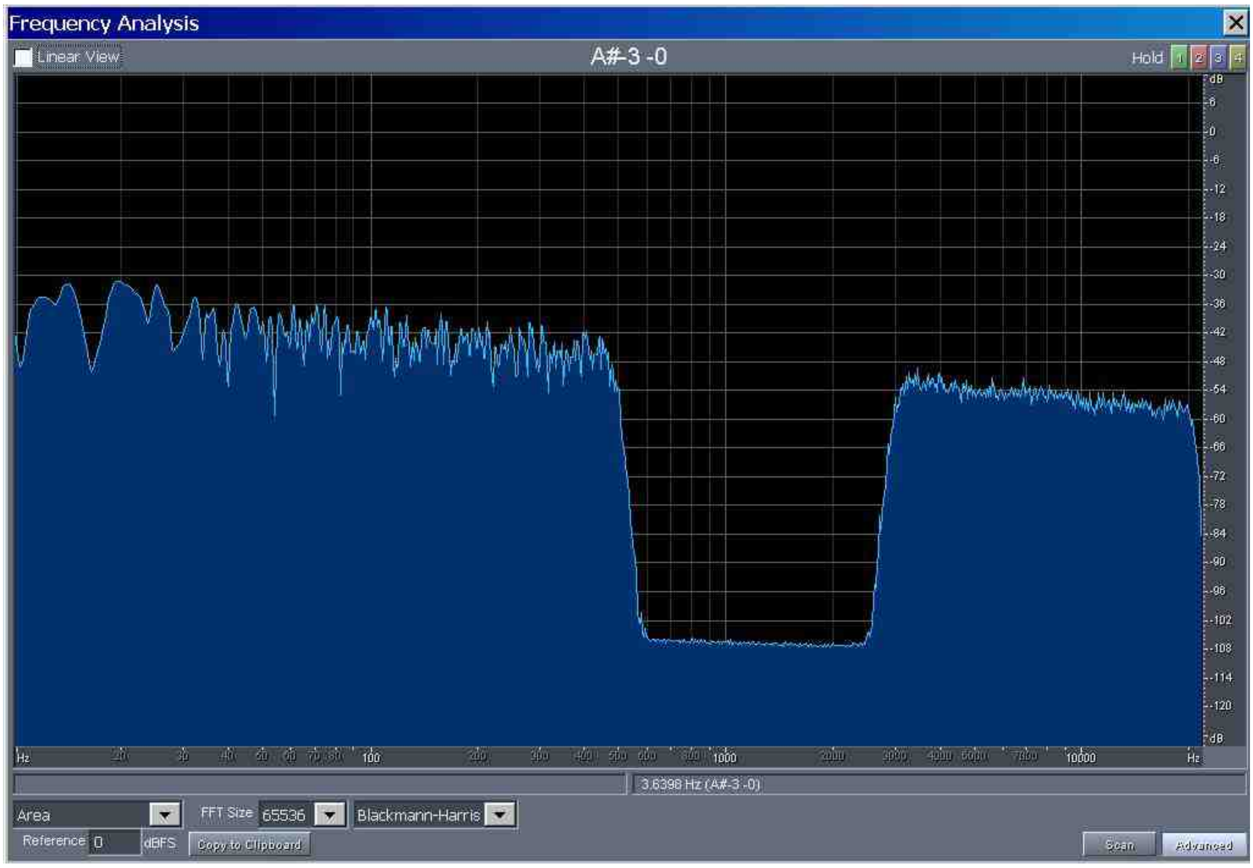

Frequency Analysis
Linear View
Hold
399.7 Hz, -118 dB
0.44069 Hz ()
Area
FFT Size 65536
Blackmann-Harris
Reference 0 dBFS
Copy to Clipboard
Scan
Advanced


Frequency Analysis
Linear View
G-3 -13
Hold
3.0394 Hz (G-3 -13)
Area
FFT Size 65536
Blackmann-Harris
Reference 0 dBFS
Copy to Clipboard
Scan
Advanced

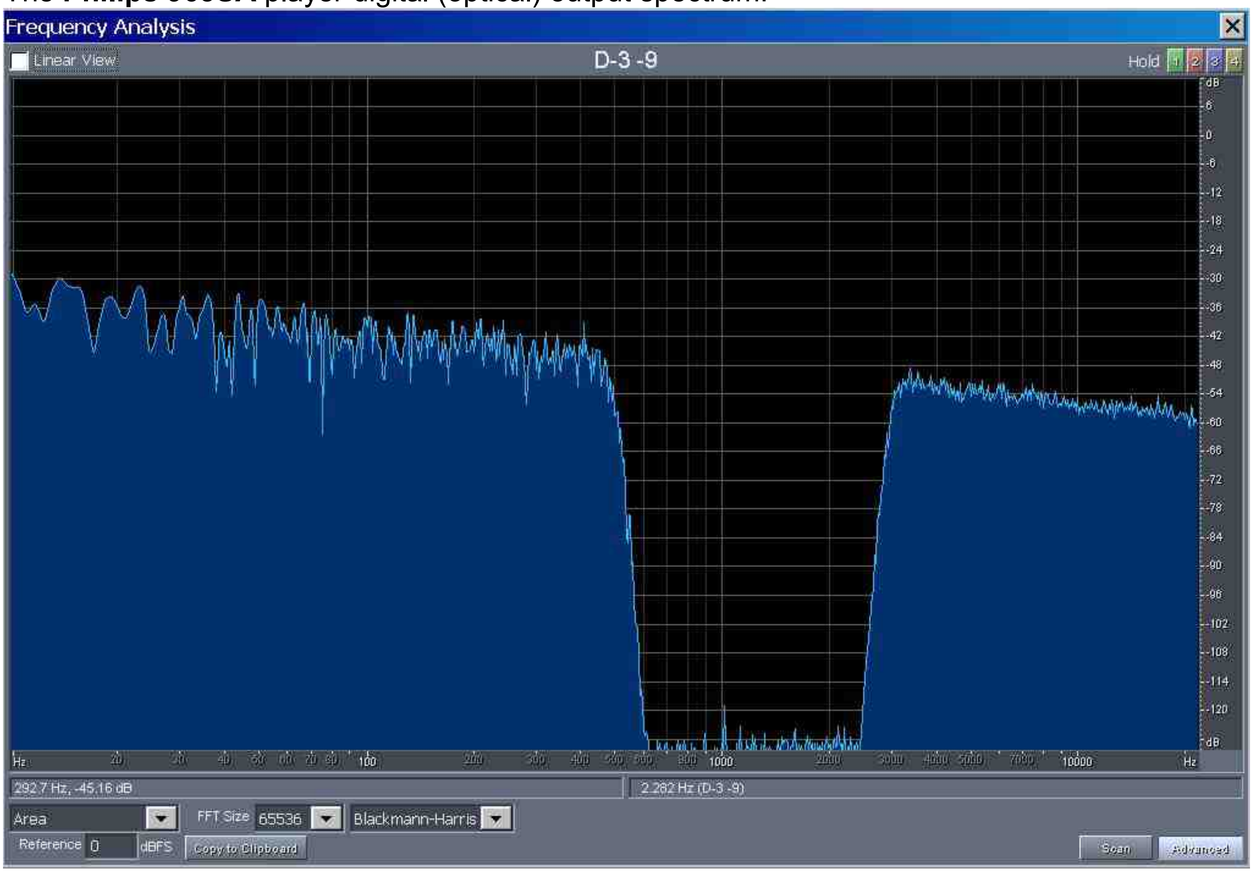
The **Philips 963SA** player analog output spectrum:



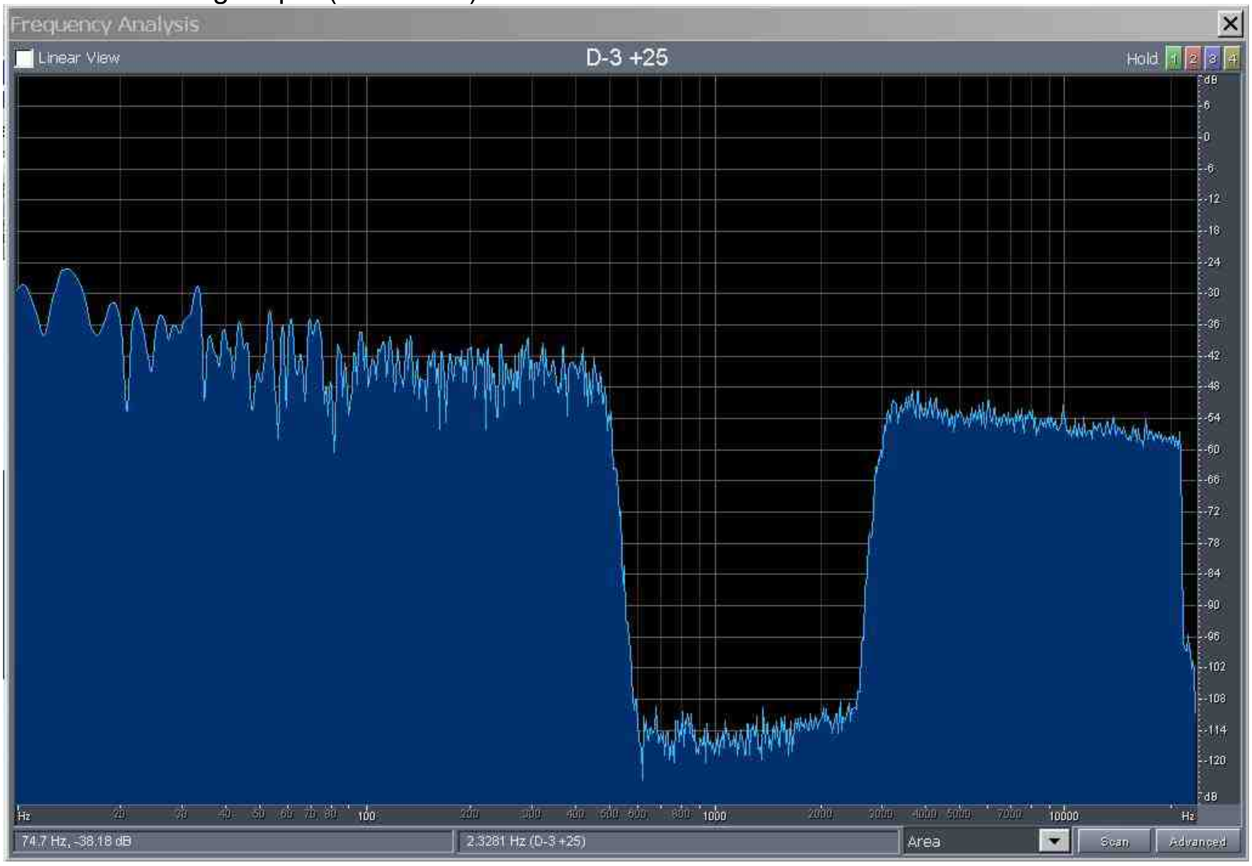




The **Philips 963SA** player digital (optical) output spectrum:



I wrote the test file in the flash memory (SD-card) of **ZOOM H4n recorder** and reproduced it via recorder's analog output (see below).



Notes:
1. Spectral component of 1000 Hz, which is visible on some graphs, is induced noise from the generator, built in my sound card E-MU 0404.
2.The TOSlink (S/PDIF) cable length is 5 meters, the price of $ 12.
3. The virtual absence of spurious spectral components in the transmission of the digital sound signal due to the high stability (less than ±10 ppm) oscillators in Philips 963SA player and sound card E-MU 0404 that provides isochronous interface [3], eliminating the operation of the PLL on the receiver.

(to be continued)

References

1. Danilov, A.A., Precision Audio Amplifiers, Hot line - Telecom, Moscow, 2004 [in Russian]
2. Belcher, R.A., A new distortion measurement, Wireless World, May 1978, pp.36-41
3. Danilov, A.A., Converters and interfaces for high-resolution digital sound formats. Electronic components, #2, 2005 [in Russian]