RightMark Audio Analyzer

The project of iXBT.com / Digit-Life

http://audio.rightmark.org

User's manual

Contents

About the program

RightMark Audio Analyzer is an independent audio measurements open-source project developed by iXBT.com / Digit-Life team. More information is available on the official site of the project http://audio.rightmark.org.

The test suite performs various tests of electro-acoustical performance of sound cards and other real-time audio devices. Testing is accomplished by playing the test signals and analyzing them after they pass through the testing chain.

System requirements: CPU: Pentium or higher, operating system: Windows 95/98/2000/XP/NT.

Preparing to the testing and testing modes

RightMark Audio Analyzer (RMAA) can be used for testing of different parts of a sound card, as well as for testing of many other audio devices. Several most usual program setups include:

- **Testing of a sound card's output (quality of playback)**. For such test an additional high quality reference sound card is needed for recording of test signals. The output of evaluated sound card is connected to the input of the reference sound card. RMAA plays the test signals through the output of evaluated sound card, and analyses the resulting signal recorded through the input of a reference sound card. It is assumed that the reference card does not introduce additional distortion to the signal (compared to the evaluated sound card).
- **Testing of a sound card's input (quality of recording)**. For such test an additional high quality reference sound card is needed for playing the test signals. The output of the reference sound card is connected to the input of evaluated sound card. RMAA plays the test signals through the output of reference sound card, and analyses the resulting signal recorded through the input of evaluated sound card. It is assumed that the reference card does not distort test signal during playback (compared to the distortions of evaluated sound card's input).
- **Testing of a full soundcard path (total recording + playback distortion)**. This test does not require any additional gear. The only requirement is that tested sound card must support full-duplex operation mode (i.e. must be capable of a simultaneous playback and recording of sound). Before testing the output of the sound card (e.g. "line out" or "spk. out") must be connected to the input of the sound card (e.g. "line in"). The drawback of this testing method is that one cannot definitely say the cause of measured distortions. They can produced either by output or by input of a tested sound card.
- **Testing of digital inputs and outputs of a sound card**. Unbelievable but fact: that many sound cards distort the signal during transmission through digital interfaces. For testing of digital interfaces one can use the techniques similar to ones described above for analog interfaces.

- **Testing of external real-time audio devices**. For such test an additional high quality reference sound card is needed for playback and recording of test signals. The output of the reference sound card is connected to the input of evaluated device, and the output of evaluated device is connected to the input of the reference sound card. RMAA passes the test signals through the evaluated device (playback and recording are accomplished by the reference sound card), and analyses the resulting signal. It is assumed that the reference card does not distort test signal during playback and recording (compared to the distortions of evaluated external device).
- **Testing of other devices (e.g. tape recorder or DAC of CD player) in asynchronous testing mode**. For testing of other audio devices RMAA has a special asynchronous testing mode. It allows outputting the test signal to the WAV file, passing WAV file through the testing devices (possibly – non-real-time), and analyzing the resulting sound from the recorded WAV-file. Here are 2 examples of using asynchronous testing mode:
    - **Testing of CD player's DAC**. RMAA generates WAV file with the test signal. This WAV file is recorded on audio CD. It is played by evaluated CD player and recorded by the reference sound card to another WAV file. This WAV file is analyzed by RMAA.
    - **Testing of a tape recorder**. RMAA generates WAV file with the test signal. This WAV file is played by the reference sound card and recorded on a cassette. Then cassette is played back, and test signal is recorded by the reference sound card to WAV file. This WAV file is analyzed by RMAA.

Running the program

Run the RightMark Audio Analyzer. If you run it for the first time, you can be prompted to select your sound card settings (audio device, sampling rate, and resolution): see fig. 1. The program will store these settings in the registry.

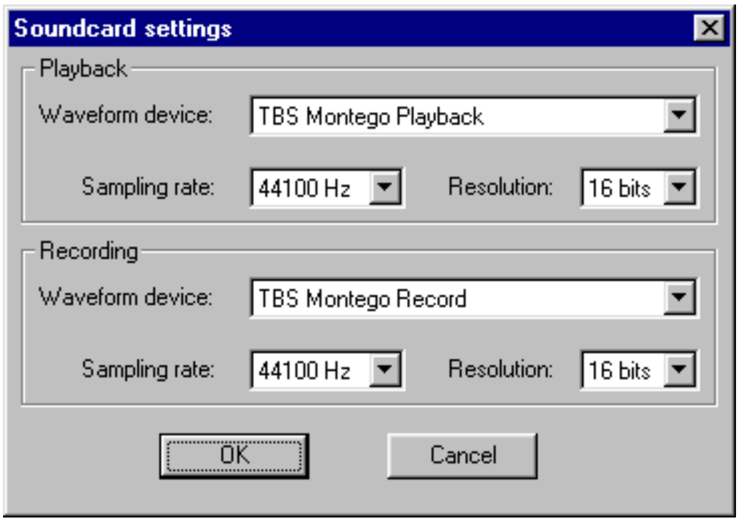

Fig. 1: *Sound card settings dialog.*

"Wave mapper" is virtual sound device that stands for the current active wave device. It can be set up in Control Panel / Multimedia section.

Main window

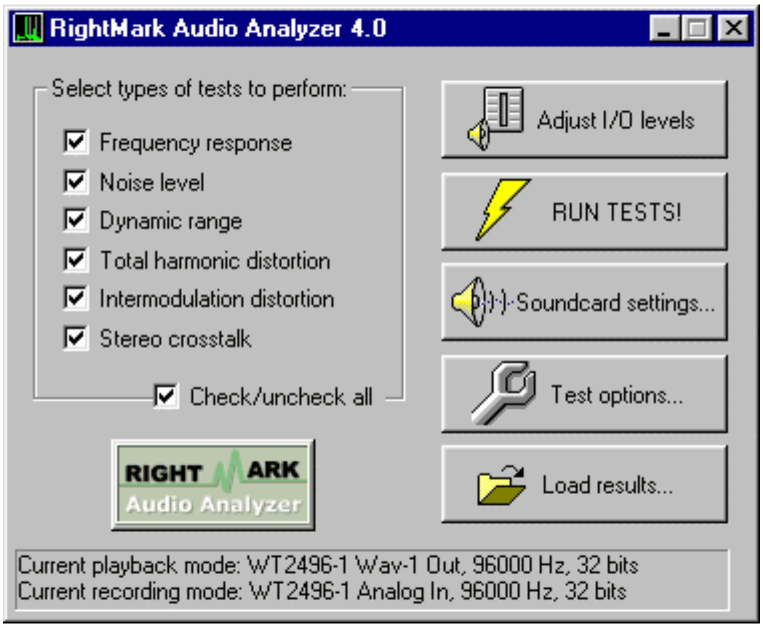

Fig. 2: *The main window of RMAA*

Adjusting levels

It is evident that playback and recording levels hugely affect the performance of audio systems. That's why proper levels setup is crucial for getting best measured results from your system. You can perform tests several times and adjust levels to get the best results.

Let's outline the process of setting levels before testing of a full sound card path (DAC + ADC), when "spk out" is connected to "line in":

1. Using the mixer of your sound card select only "wave out" and "master" sources for playback, and only "line in" source for recording. Turn off all equalizers, tone controls, 3D surrounds, etc. to obtain the most adequate test results.
2. Set "wave out" level for playback and "line in" level for recording to their default positions (usually somewhere from middle to upper positions).
3. Run RMAA program and setup the *"Soundcard settings"*. After that press the *"Adjust I/O levels"* button.
4. A sound card calibration will start. Two testing signals will be played back repeatedly: 0 dB FS and -6 dB FS. Adjust playback and recording levels in your mixer to make input levels approximately equal to output levels (precise equality is not needed, but it is desirable for accuracy; a difference of 1 or 2 dB can be tolerated). It is recommended to start adjusting levels using "master out" slider only in your mixer. If it does not lead to success, try "wave out" and "line in" sliders. At the spectrum windows you can watch spectra of incoming signals. You should maximize the amplitude of 1 kHz harmonic, and prevent huge harmonic distortion (fig. 3).

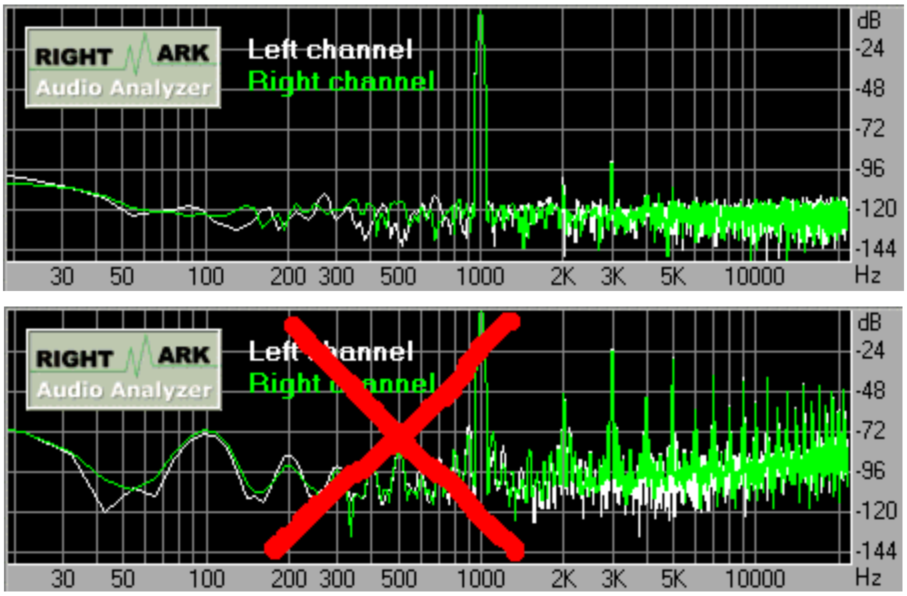

Fig. 3: *Adjusting levels. Correct and incorrect setups.*

Running the tests

Select the desired types of tests at the main window (see fig. 2) and press the *"RUN TESTS"* button. All the tests take about 30 seconds to complete. After tests are complete you can view the results in the "Test results" window or perform tests that were not selected.

Viewing the results

In the "Test results" window you can see the information on results of all the tests that were performed (see fig. 4).

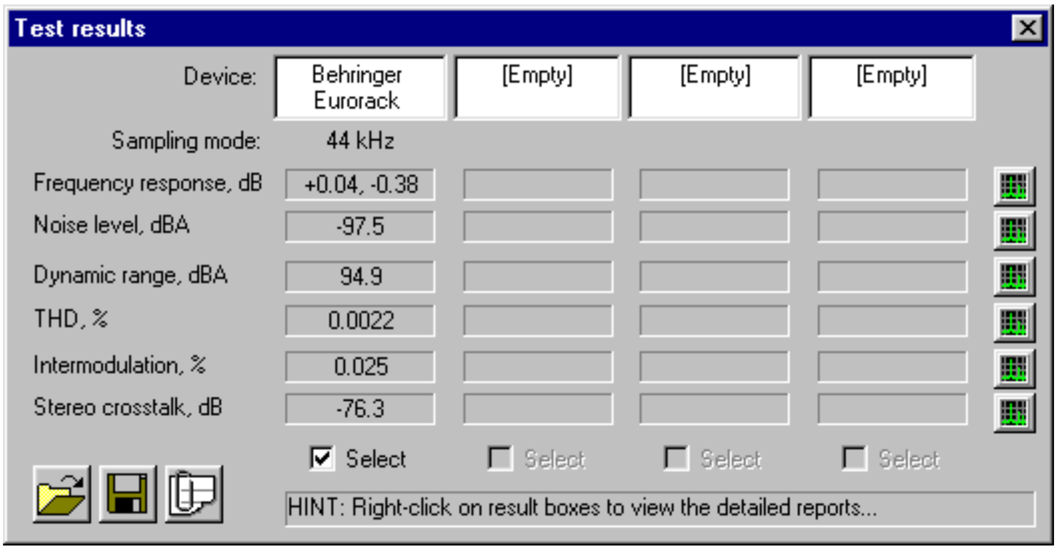

Fig. 4: *"Test results" window*

The window is divided into 4 slots (4 columns), each capable of storing one set test rests. You can load 4 sets of results simultaneously and compare them.

For each test a brief numeric result is displayed. You can get a more detailed report by right-clicking on a corresponding numerical result.

A column of buttons to the right from result slots enables watching spectrum graph for corresponding test.

"Select" buttons at the bottom of each slot select several slots for comparison of results.

Buttons of opening and saving enable loading results from SAV file or saving results to SAV file. SAV file stores all the detailed reports as well as spectrum graphs.

The button of HTML report generation enables to generate a HTML file with test results or with comparison of test results from several slots. HTML report includes all the detailed reports and spectrum graphs.

Graph window

Let's look at the Graph window

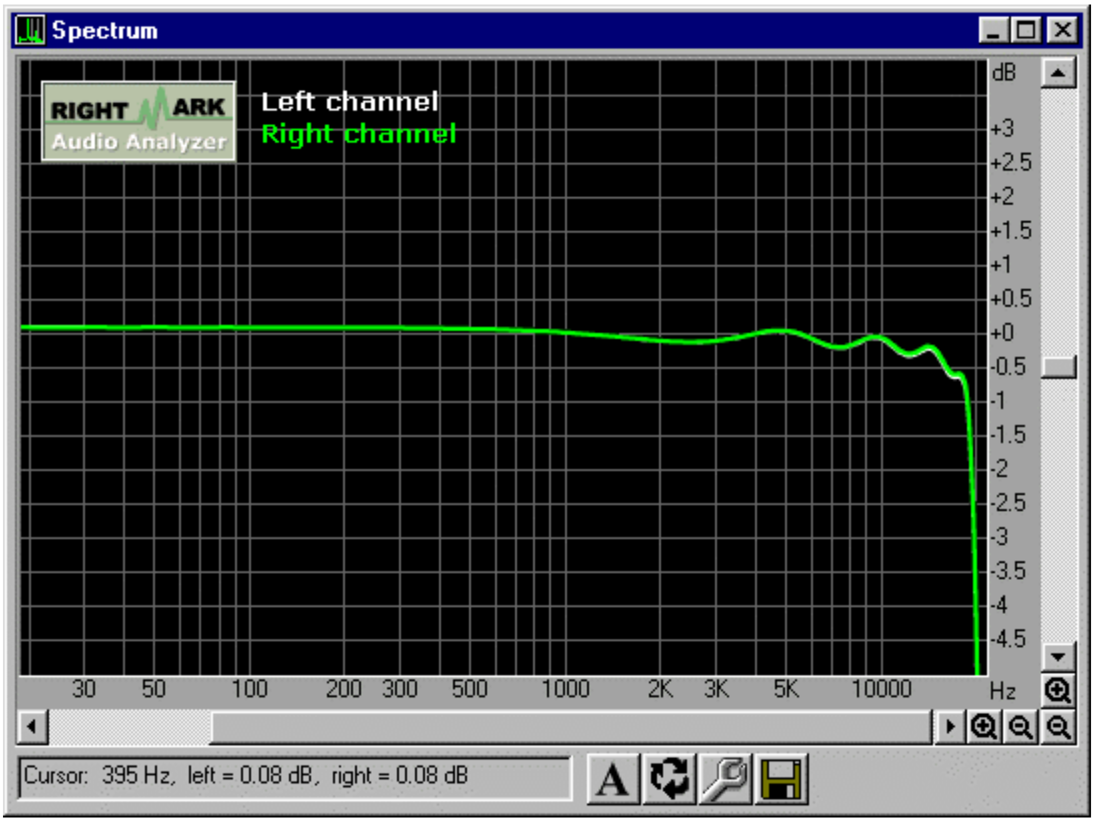

Fig. 5: *Graph window (for frequency response test)*

There are the following control buttons here:

 - zoom in

 - zoom out

Toolbar contains the following buttons:

- Anti-aliasing of a graph – smoothes the graph output on a screen (eliminates the effect of "jagged edges").
- Swap stereo channels – draws the graph for left channel in front of the one for right channel (default setting is vice versa).
- Setup of graph parameters.
- Saving graph in the PNG file.

Mouse controls:

Left button – selects the horizontal range and performs "zoom in".

Right button – performs "zoom out".

Tests description

Please see the description of tests at the separate manual (TESTS.PDF file) or online.

Test options

This dialog selects some options for various tests:

- *Asynchronous testing mode* checkbox turns on asynchronous testing mode when test signal are played and recorded not through a sound card but save to the WAV file or analyzed from WAV file.

Soundcard settings

This dialog selects devices for digital audio playback and recording, and mode of operation: sampling rate and resolution.

In this version of the program the sampling rates for playback and recording must be equal. Audio devices and resolutions may differ.

After closing the dialog with the *"OK"* button the program tries to test selected devices and the selected modes of operation and reports on failure.

Glossary

A-weighting. Human hearing is unequally sensitive to sounds of different frequencies. For example the maximum of our sensitivity to quiet sounds lies around 3 kHz. Sounds of these frequencies we perceive as louder ones. Because of this we need to modify the technique of spectral measurements to make them closer to our hearing perception. Such modifications are known as A-weighting. They are widely used in audio measurements (for example, when estimating the noise level or dynamic range). As a result of A-weighting we get inaudible frequencies attenuated, while the most audible ones contribute more to the final results.

Dynamic range is a ratio of the maximal signal amplitude to the noise level (RMS value) in the presence of weak signal.

Dithering is a process of adding noise to the signal which was generated with a high precision. Usually dithering is performed before quantization to eliminate correlation between the signal and quantization error. As a result the noise of quantization becomes white and more pleasant than the "dirty" quantized signal. Although we add some noise, the dynamic range becomes subjectively higher.

Full-duplex mode is an ability of sound card to record and playback digital waveform audio simultaneously. Almost all of modern soundcards support this mode, but some of them need to be tuned up in the Control Panel.

THD (total harmonic distortion) is a level of (usually unwanted) harmonics generated in the sound device. Usually high quality devices have a low THD value (lower than 0.002%), but there are exceptions. Many tube devices have rather high THD level, which makes their sound “warm”. But transistor devices must have low THD, because their (odd) harmonics don’t make the sound pleasant.

Feedback and future work

In the next versions of this project we plan to make extend our online database for comparison of different sound cards. The set of tests will be extended.

All your comments and suggestions on this program will be highly appreciated. The best place to contact us is a forum on our official site. Please contact us on any questions or problems you experience with RightMark Audio Analyzer.

http://audio.rightmark.org – the official site of the project,

lukin@ixbt.com – Alexey Lukin, author and programmer of RMAA,

maxim@ixbt.com – Maxim Liadov, RMAA project manager,

http://www.ixbt.com

http://www.digit-life.com