RightMark 3DSound
v 1.2
User's Manual

Introduction

There are many PC sound devices in the market: integrated AC’97 and HDA codecs and dedicated chips, integrated hardware and software controllers, soundcards supporting software, hardware and combined sound buffers processing, external multimedia USB and IEEE1394 devices.

Every manufacturer declares (and users demand) specific device features. For example, support of particular APIs and extensions, specific number of hardware accelerated sound buffers. At that it's impossible or hardly possible to check the presence and quality of features declared for any sound device or new drivers without special software.

Therefore, a dedicated sound test is required to diagnose and test the feature implementation quality of specific sound system.

Test Purposes

The idea of RightMark 3Dsound is an independent synthetic sound test for testing hardware features and quality of DirectSound-compatible devices software support.

Application Range

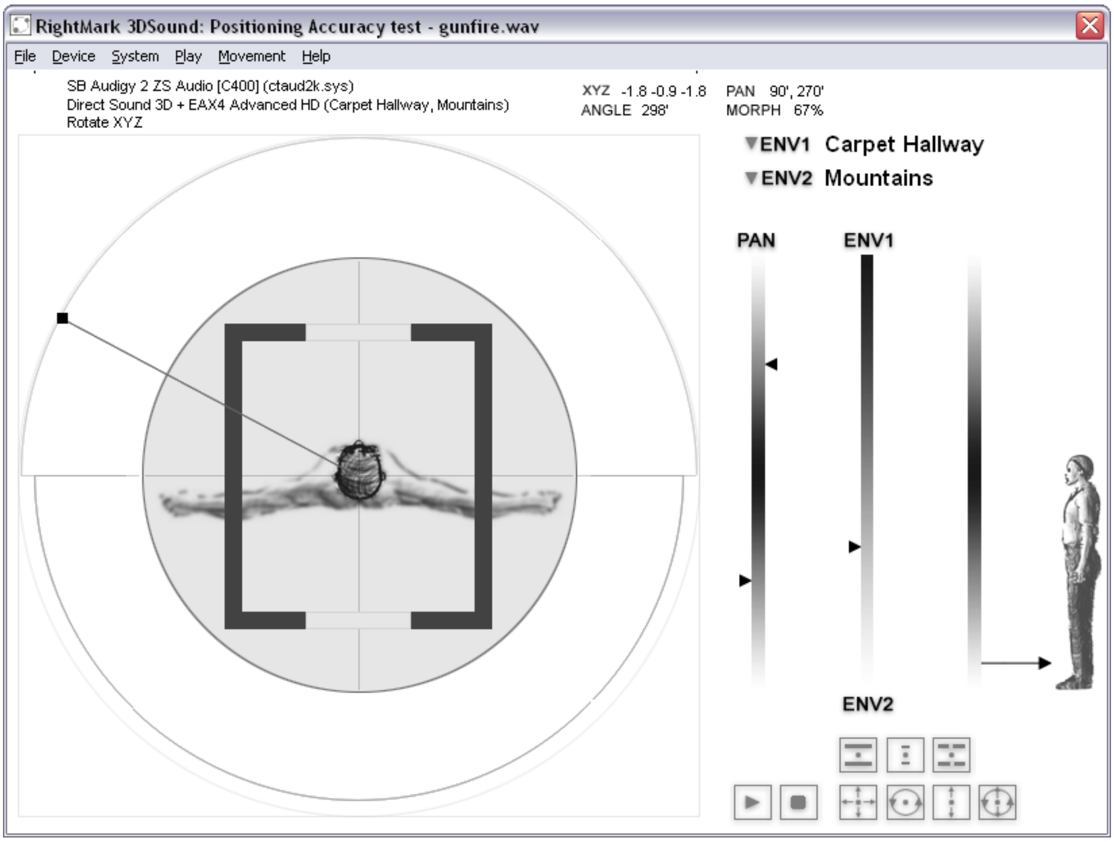
Operating systems: Windows 98SE/ME, Windows 2000/XP.
Sound APIs: DirectSound, DirectSound3D

Test Suite

- RightMark 3DSound: Positioning Accuracy test
- RightMark 3DSound: CPU Utilization test
- RightMark 3DSound: Data Analyzer

RightMark 3DSound: Positioning Accuracy test

This component tests DirectSound3D sound source positioning accuracy. The test implies subjective listening to a 3D scene with a sound source moved in 3D space by a user. This test supports all main features of EAX4 Advanced HD.



Environment filtering (EAX3, EAX4)

Obstruction

The direct-path sound wave can only reach the listener by transmission through the obstacle or by diffraction around the obstacle. In both cases, it will be partially or completely muffled. That muffling effect is called obstruction.

Occlusion

Source and listener are completely separated by a wall so there is no direct air connection between them. Any sounds that pass from source to listener must pass through the wall, which muffles the sounds. This is called occlusion.

Exclusion

We can now make an opening in the wall that separated listener from source, for example a doorway. Source and listener are still separated by the wall, but there is an opening allowing the sound to enter the room and the direct path between the source and listener is clear. When there is a direct path, this is called exclusion.

Environment Reflections (EAX3, EAX4)

Knowing a little bit about the architecture of a room as well as the position and orientation of the listener, it is possible for a program to calculate and set the direction from which the early reflections tend to come from (for instance, by considering the positions of the nearest walls relative to the listener).

Environment Panning (EAX3, EAX4)

Alternatively, imagine a situation where the listener is in a non-reverberant space, but hears reverberated sounds coming from the mouth of a nearby cave. The program can pan the reflections and the reverberation so that they are perceived as being emitted from the cave opening.

Multi-environments (EAX4 only)

An EAX 4.0 multi-environment implementation can provide a more realistic experience by enabling the user to hear acoustic information from rooms other than the one occupied by the listener.

Environment Morfing (EAX3 partially, EAX4 in full)

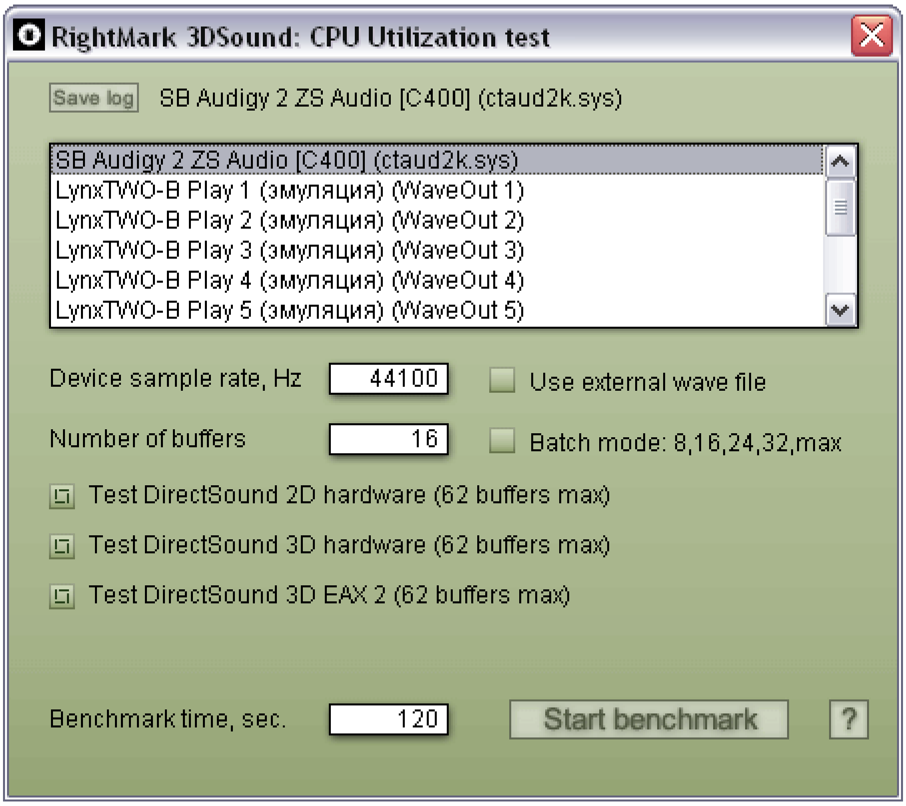
The environment panning feature has the added effect of allowing EAX 4.0 multi-environment implementations to handle in a very natural manner the transition that occurs when the listener moves from one environment to another. And listener hears all presented environments.

When we are moving between 2 different environments in the EAX3 test mode, parameters of the first preset are linear interpolating to the second ones. But we have just the same single environment with changing parameters.

RightMark 3DSound: CPU Utilization test

This synthetic test is used to measure CPU utilization depending on operation mode of the selected DirectSound device. The test emulates the primary cycle of a typical game sound engine. The results are obtained through a special CPU utilization measurement algorithm getting data in specific intervals (0.5 s) that provides a higher degree of accuracy and lower dispersion for smaller CPU load than WindowsXP standard system counter. To control background CPU utilization a measurement is conducted in the beginning of every test before any actions are performed.
Result of the CPU Utilization test is average value and deviation, highlighted in Data Analyzer.

Attention! We strongly recommend you to stop all unnecessary applications and services (Start/Settings/Control Panel/Administrative Tools/Services) before you run the test.
The test length should not be shorter than 120 s for consequent statistics purposes.
If you don't provide these conditions test developers do not guarantee results accuracy.



Clicking **Save log** you can save the file containing feature diagnose of the selected device. Besides the standard DirectSound diagnostics, the program checks for EAX support.

Sample rate: By default it's set to 44100 Hz, which is the standard for modern games, so you shouldn't change it. The value given is the same for DirectSound Primary Buffer and Secondary Buffers. This is done to prevent buffer sample rate conversion that might add to CPU utilization (depends on drivers and feature implementation).

Use external wave file: loads «rightmarktestfile.wav» from the current folder. This file must be in mono format and at the same synchronizing frequency as in test. If you don't use an external wave file, the test will automatically generate it mathematically. Sound generation is borrowed from PC graphics, where «X xor Y» texture is often used for tests. It looks like a fractal and sounds unusual. We believe it's better than just a sinusoid signal or noise. For the synthesized sound the secondary buffer size is 256*256*4 = 262144 bytes.

Number of buffers: *Buffer* is a single sound stream in terms of DirectSound API. It doesn't affect the batch mode. You can set the amount of buffers before the test. In Direct Sound software mode amount of software buffers is limited to 128. The amount of hardware buffers is limited only by driver capabilities of the sound device. If the buffers amount set exceeds the available amount, it's automatically truncated to the latter.

Batch mode: enables to run 5 standard tests for a selected sound device one after another varying only the amount of buffers (8, 16, 24, 32, max).

Test DirectSound… (n buffers max): enables to select tests for separate testing. Doesn't affect the batch mode.

Before the test is run, the program creates a separate buffer (typical for most games) for mixing (in case of fully software buffer) or special parameters setting (position, volume, EAX source properties) for DirectSound3D buffers for DirectSound hardware mixing.

As parameters are not updated all the time, buffer data is executed each 50 ms. In a buffer a procedure is executed that randomly changes position of each sound source (static buffers) every 50 ms within 2-meter range from the listener (camera). To fully simulate the game cycle, listener (camera) position and orientation changes within 1-meter range as well.

Another important gaming parameter is **Transfer Rate**, which is time of loading a sound wave data into a static sound buffer. In our test we measure time of loading 32 buffer data into DirectSound3D buffer to obtain transfer rate in MB/s. (Note that for software buffers load is less by order than for hardware buffers.)

An example of DirectSound diagnostics log

Device: SB Audigy 2 Audio [8400] (ctaud2k.sys)

Features:
DirectSound 3D Hardware: Yes
DirectSound 2D Hardware: Yes
EAX 1: Yes
EAX 2: Yes
EAX 3: Yes
EAX4 Advanced HD: No

Rates:
dwMinSecondarySampleRate 4000
dwMaxSecondarySampleRate 192000

Free buffers stats:
dwFreeHw3DAllBuffers 62
dwFreeHw3DStaticBuffers 62
dwFreeHw3DStreamingBuffers 62
dwFreeHwMixingAllBuffers 62
dwFreeHwMixingStaticBuffers 62
dwFreeHwMixingStreamingBuffers 62

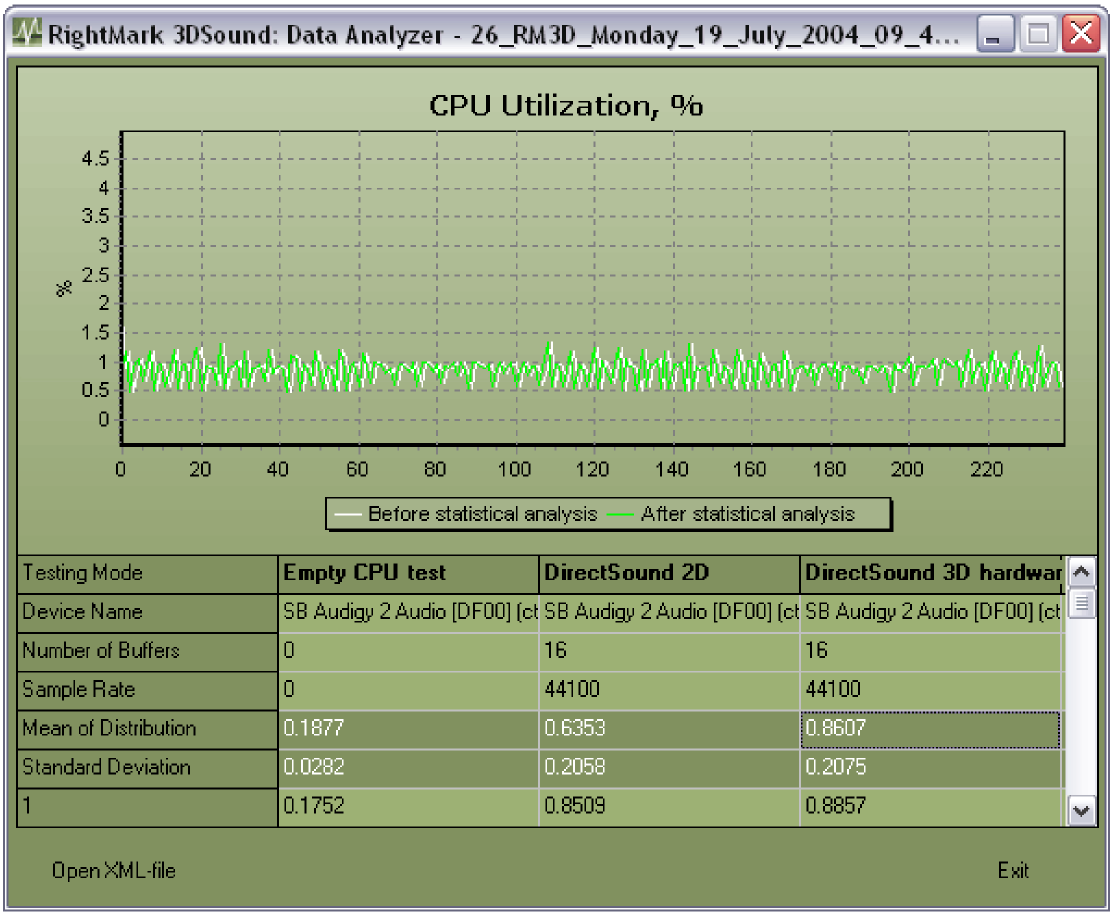
Max buffers stats:
dwMaxHwMixingAllBuffers 64
dwMaxHwMixingStaticBuffers 64
dwMaxHwMixingStreamingBuffers 64
dwMaxHw3DAllBuffers 64
dwMaxHw3DStaticBuffers 64
dwMaxHw3DStreamingBuffers 64

Misc stats:
dwFreeHwMemBytes 0
dwTotalHwMemBytes 0
dwMaxContigFreeHwMemBytes 0
dwUnlockTransferRateHwBuffers 0
dwPlayCpuOverheadSwBuffers 0

Audio transfer speed (hardware): 3.731 Mb/sec.

RightMark 3DSound: Data Analyzer
(XML parser / statistical analyzer)

To exclude the effect of random OS load peaks and to obtain more useful data on CPU utilization character we use statistical data processing. This gives us not only the mean of distribution, but also the standard deviation.
After thorough analysis of peaks and utilization distribution function we developed the special method of data filtering. It bases on a very simple idea to filter all random results that do not correspond to system regularity analyzed. If data peaks are present all the time, they won't be filtered. Therefore it's important to set long test time. As our experiments have shown, 120 s (240 data values) is the optimal time length.
Actually anyone can also read and parse XML logs and offer his own processing method. To provide visual control of data processing quality, we've implemented graph drawing at diagram before and after processing.



Data are analyzed automatically once a XML file is opened. To construct a diagram, click the needed column. To zoom in and out, select a range holding the left mouse button. To move the viewing window around the diagram, hold the right mouse button.