Why Amarra?

When the Sonic System was proposed in 1988 the world of digital audio had a very different landscape. The first Sonic System was based on the work pioneered by James Andy Moorer while at LucasFilm Ltd in the 1980's and even earlier while at Stanford in the 1970s. The initial systems cost over $45,000 and used then new 16 MHz Apple Macintosh II. The Sonic System became the first digital audio workstation designed for the restoration, editing, processing and mastering of audio for CD release. Abbey Road and MCA were two of Sonic's first customers and remain customers to this day. Over the years, our popularity has grown to include almost every major studio in the world. Many of the CDs and movies you listen to and see have been produced using a "Sonic". Today, the Sonic Studio Engine (SSE) is the tenth generation system derived from the original Sonic System in 1988. The Amarra sound is a result of the past twenty years of research into sound, sound quality, numerics and signal processing.

There are many factors involved in the proper design and implementation of a professional digital audio reproduction system. We will let others talk about the merits of USB vs. FireWire and Digital Audio Converters, suffice to say the Sonic Studio products are designed for very exacting customers who require the best for their work. That leaves us with the software to explore. There are many steps involved in the playback of audio from a computer that all computer based players must perform. These include input from the hard disk, format conversion, processing, and playback. Each is described below.

The first task is that the audio file, represented as digital data, must be read off the media and into memory. This requires the audio player to guarantee (as best it can) the audio will be read from the disk in time. The player needs to deliver the audio samples to the Hardware Interface not only on time but also synchronized. Otherwise, we may hear clicks, dropouts, phase, and balance problems. Sonic addresses this using optimized file handling routines. We do not look at latency as an issue in a home audio reproduction system unless it's being synchronized with video.

Once the audio is in memory, it needs to be converted to a format the computer can "understand". Current audio formats for uncompressed audio use a 16 or 24 bit sample and this must be converted to the IEEE floating point architecture in use by most computers today. This conversion can introduce noise into the audio signal. Should we truncate, round, scale or perform some combination to achieve the best sounding result? This conversion occurs on input from the disc and on output to the hardware interface. Based on our experience, we sometimes find that textbook math does not mean the best sounding math.

The next stage is the gain structure and processing that may take place inside the audio engine. When you adjust the slider to control the volume a gain process is applied to the audio signal. Even when the gain is at full volume processing may take place that effects the sound. In the Sonic Studio Engine we perform all calculations using 64 bit extended floating point math. This allows us a full 56 bit mantissa which allows us to keep the noise levels below the 24th bit found in high resolution audio. When any processing is applied it is important to redither the sounds to mask the effects of all the rounding and noise that were previously introduced. To address this problem, SSE comes with two dither algorithms which are optimized for use with Amarra. Perhaps as important as the underlying math is the efficiency in implementation as each operation has the possibility of increasing the noise.

As a closing example, lets look at one set of measurements of Amarra versus iTunes. When playing a 24 bit audio file iTunes had a noise floor (FFT) at about -110 dBFS with a full scale audio input. For 24 bit audio this is noticeable and definitely not good for high end reproduction. Even with the iTunes volume at Full Gain, the signal is attenuated by about 2.5 dB. On the other hand, using Amarra showed a noise floor below -128dB with no attentuation when at 0 dB.

In addition, Amarra can take advantage of many features commonly found only in professional workstations. As we move forward we can look forward to bring the highest quality tools for Sample Rate Conversion, CD Ripping, CD Burning, Vinyl Restoration, and much more. Just one more bonus of 20 plus years of development.

Measurements:

The following graphs illustrate the differences between Amarra and iTunes. The test was performed using Amarra 1.0 and iTunes 8.1.1. The "sound enhancement" feature of iTunes is disabled, and the Sample Rate is set prior to running the test.

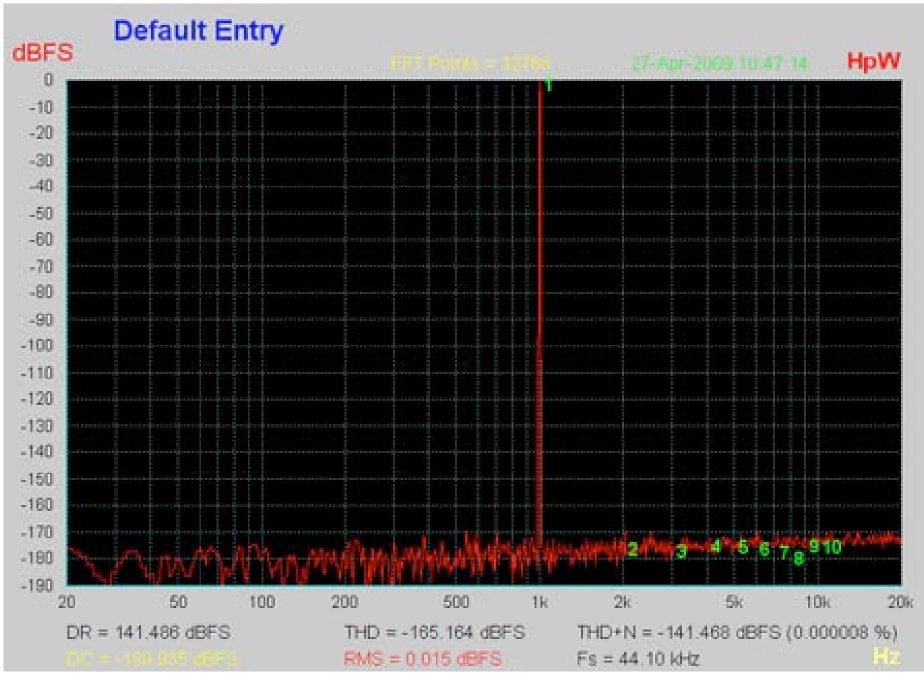

Figure 1.
24 bit 0dBFS sinewave at 44.1kHz played back with Amarra through the Minerva via its digital output.
The D/A converter is not used. You see in that FFT what you would expect for a 24 bit sinewave.

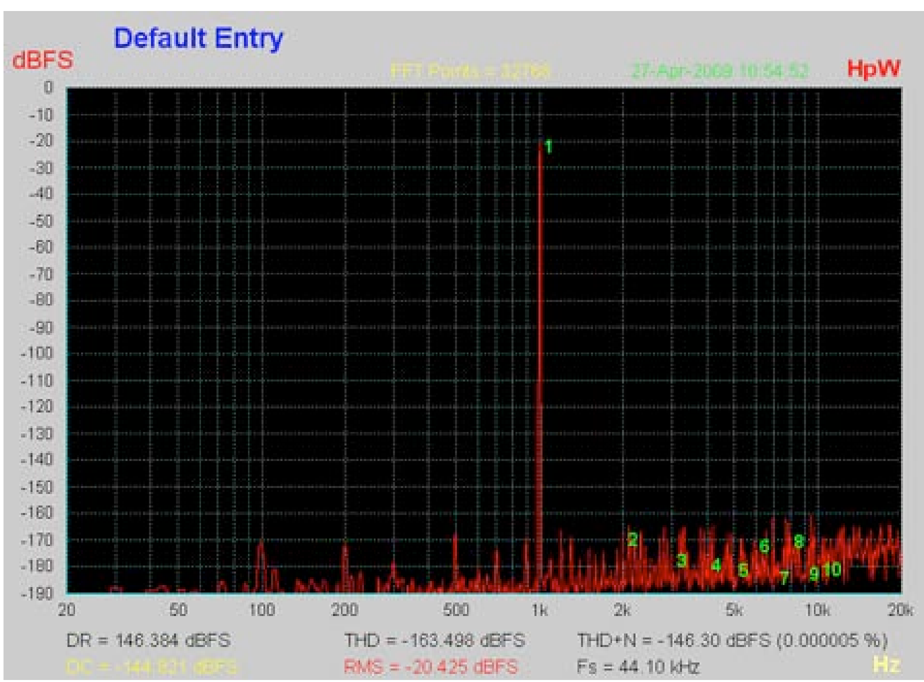

Figure 2.
This is the same signal with the gain lowered by about 20dB via the Amarra fader.
Note the quantization distortion due to dither lacking, which was not enabled for this measurement.

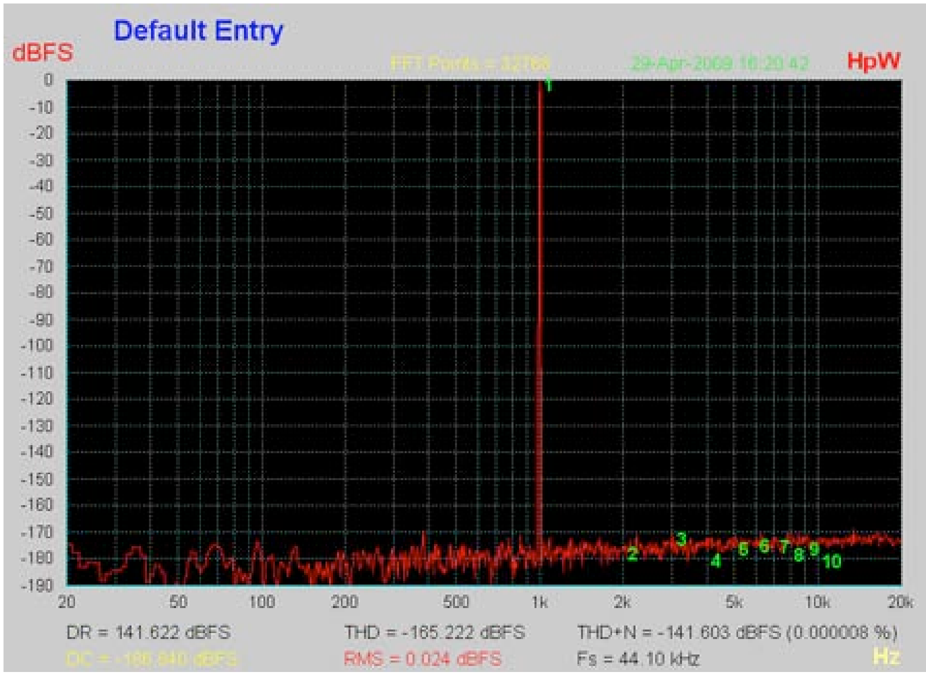

Figure 3.
This is the same sine wave played with iTunes.
This is with "sonic equaliztion" disabled.

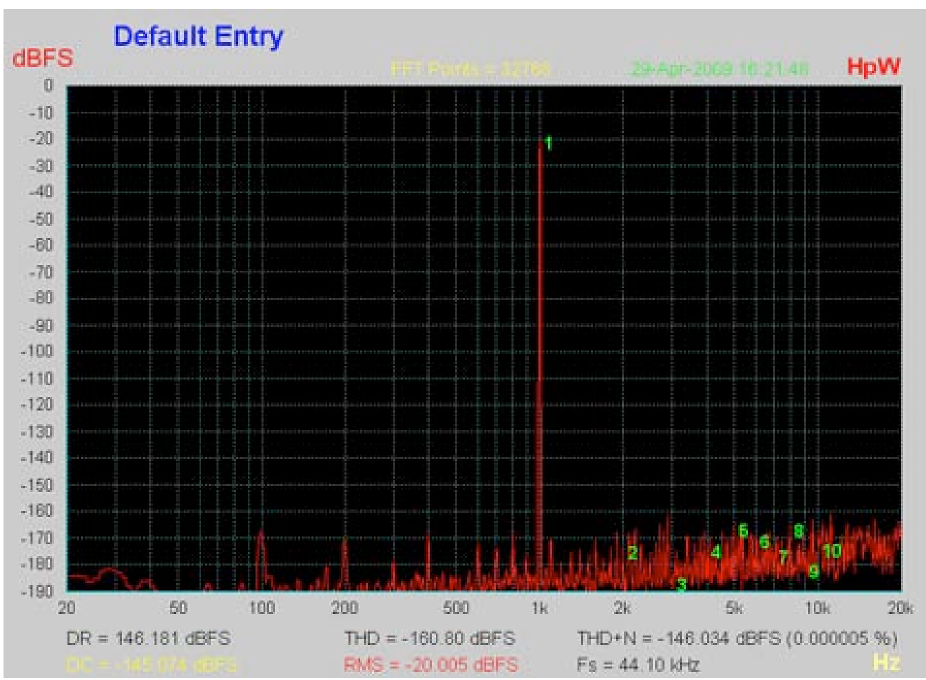

Figure 4.
The same signal at about 20 dB lower level, done with the iTunes fader.
Graphs courtesy of Weiss Engineering Ltd.